YX GoIP

User Manual

V4.6

YX GoIP16-16

YX GoIP16-64

YX GoIP16-128

YX GoIP32-32

YX GoIP32-128

www.yx19999.com; www.szyxtel.com
TEL: +86-4007770752; +86-752-2777557 FAX: +86-752-2159233
Email: support@yx19999.com
Address: 2-2308 Yixin Building, No.11 Huabian South Road, HuiZhou, GuangDong, China
zip code: 516000

CONTENT

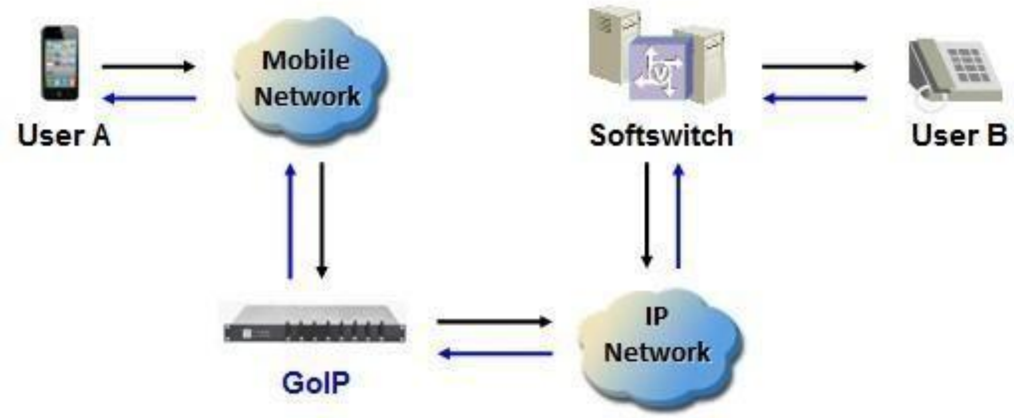
❖1 Introduction

1.1 Overview

A VoIP GSM Gateway (GoIP Gateway) is a device which reduces costs when calling from a fixed telephone line to mobile network. It enables direct routing between IP, digital, analog and mobile networks.

GoIP Gateway is now used more and more for telephone carriers to land their IP calls to mobile network. In those areas where fixed line services are unavailable or much more expensive than the mobile cost, GoIP Gateway is an irreplaceable alternative.

The following figure shows a basic topology of GoIP Gateway usage.



1.2 Glossary

- VoIP: Voice over Internet Protocol.
- SIP: Session Initial Protocol.
- DTMF: Dual Tone Multiple Frequency.
- IMEI: International Mobile Equipment Identity (with15 digits).
- ASR: Answer Seizure Ratio.
- ACD: Average Call Duration.
- PDD: Post Dial Delay.
- LCR: Least Cost Routing.
- USSD: Unstructured Supplementary Service Data. GSM: Global System Communications.
- CDMA: Code Division Multiple Access.
- WCDMA: Wideband Code Division Multiple Access.

❖2 Equipment Information

2.1 Product Brief

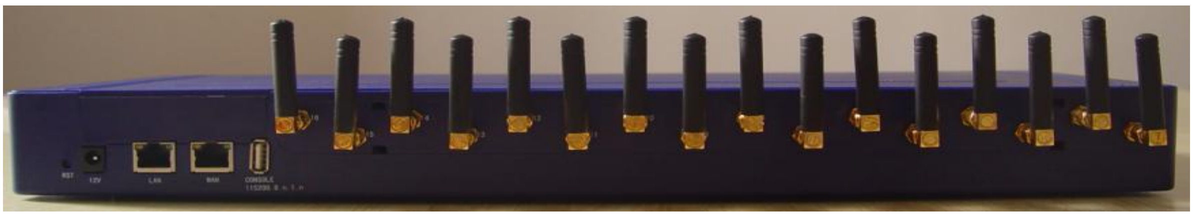
YX GSM/CDMA/WCDMA series VoIP Gateway is a multi-functional and high performance product, which is designed with advanced embedded technology. YX series is able to process traditional voice call service and internet data service. perfectly support G729a/b/e、G723.1、G.711 a/u law and iLBC codecs at the same time.

YX series VoIP Gateway please check the following table about the difference:

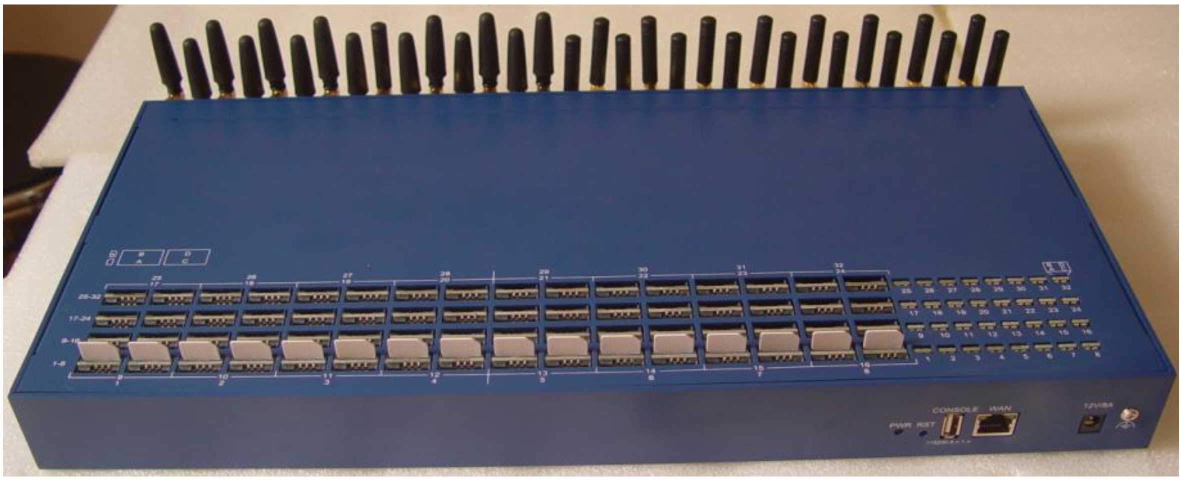
Model Number	Channel	Sim capacity in each channel	Total sim capacity
YX 16-16	16	1	16
YX 16-64	16	4	64
YX 16-128	16	8	128
YX 32-32	32	1	32
YX 32-128	32	4	128

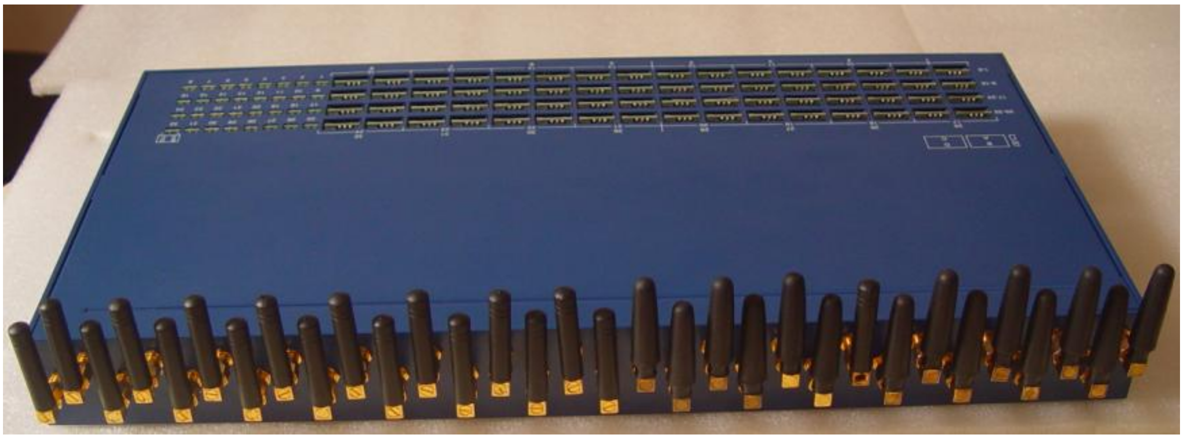
2.2 Appearance

YX16-16 and YX16-64



YX32-32 and YX32-128





- 16 or 32 Antennas
- 1 USB Serial Port (Baudrate 115200)
- 2 Network Interface (RJ45)
- 1 Power Interface (DC 12V 5A or 8A)
- 1 Reset Button
- 16/32/64/128 sim card slots
- 16/32/64/128 LED lights

2.3 Special Features

- Support G729a/b/e,G723.1,G.711 A/U law, iLBC auto-selecting
- BO(Bandwidth Optimization)
- Proxy Encryption Solution for IP Block
- Support SIM Bank
- VPN(pptp)
- SIM Card Rotating(YX16-64/YX16-128/YX32-128)
  - Solution to solve the SIM card blocked by:
    - Accumulated Call Duration
    - Accumulated Connected Calls

- ✧ Accumulated Calls
- ✧ Consecutive Failed Calls
- ✧ Consecutive No-Alert Calls
- ✧ Consecutive No-Answer Calls
- ✧ Consecutive No Carrier Calls
- ✧ Consecutive Short-Duration Calls

- Station intelligent switching
- RMS(Remote Management System)
- Port Inter-Calling
- Fake Ringback
- Call Duration Limitation for SIM Card/Single
- Dial Plan/Prefix Inward Translation/Intelligent Routing
- Web Browser: Firefox/Chrome /IE/Opera

2.4 Specification

Model	YX16-16/ YX16-64/ YX16-128/ YX32-32/ YX32-128
Number of Channels	16 or 32channels
Frequency	GSM: 850/900/1800/1900MHz (full band) CDMA:450; 800; 800/1900MHz (optional) WCDMA:900/1800/2100MHz; 850/900/1800/1900MHz (optional) (YX16-128/ YX32-32/ YX32-128 not support CDMA & WCDMA)
SIP Specification	SIP/2.0 RFC3261 Session Timer RFC4028 STUN
Network Protocols	DHCP/PPPoE/VPN(pptp) NTP Telnet/HTTP/FTP/TFTP Encryption:YX,VOS2000,RC4,XOP.Base64
Telephony Features	Hot-line call ,Dial plan, Speed dial, Phone book, CDR, LCR, White/Black list
Telephony Signaling	DTMF tone dectection/generation DTMF relay: in-band, RFC 2833 and SIP info Call forward: unconditional, no answer and busy N-way conferencing

Voice Capability	Voice codecs:G729a/b/e,G723.1,G.711 A/U law, iLBC Echo cancellation Silence suppression & detection（VAD, CNG） Adaptive jitter buffer Volume adjustable
Number of Ports	1 WAN 10/100Base-T ethernet(RJ-45 connector) 1 LAN 10/100Base-T ethernet(RJ-45 connector)(YX32 not the port) 1 Console(USB)
LEDs	1 Power and 16 or 32groups of card online and running status indicator
Power Supply	100-240VAC, 50 - 60 Hz IN, 12VDC/5A or 8A Out
Operating Environment	Operating temperature: 0 - 50ºC Operating humidity: 10 – 90%RH
Warranty	12 Months

2.5 Mobile Features

- SMS Send, Receive and Forward(GSM/SIP/HTTP)
- SMS Inbox
- AT Command , USSD
- SMS Format: PDU/TXT
- PIN Code Management
- CDMA Delay Answer
- GSM Polarity Reversal
- Carrier Selection
- Caller ID Hidden(need SIM Card support)

2.6 Maintenance and Management

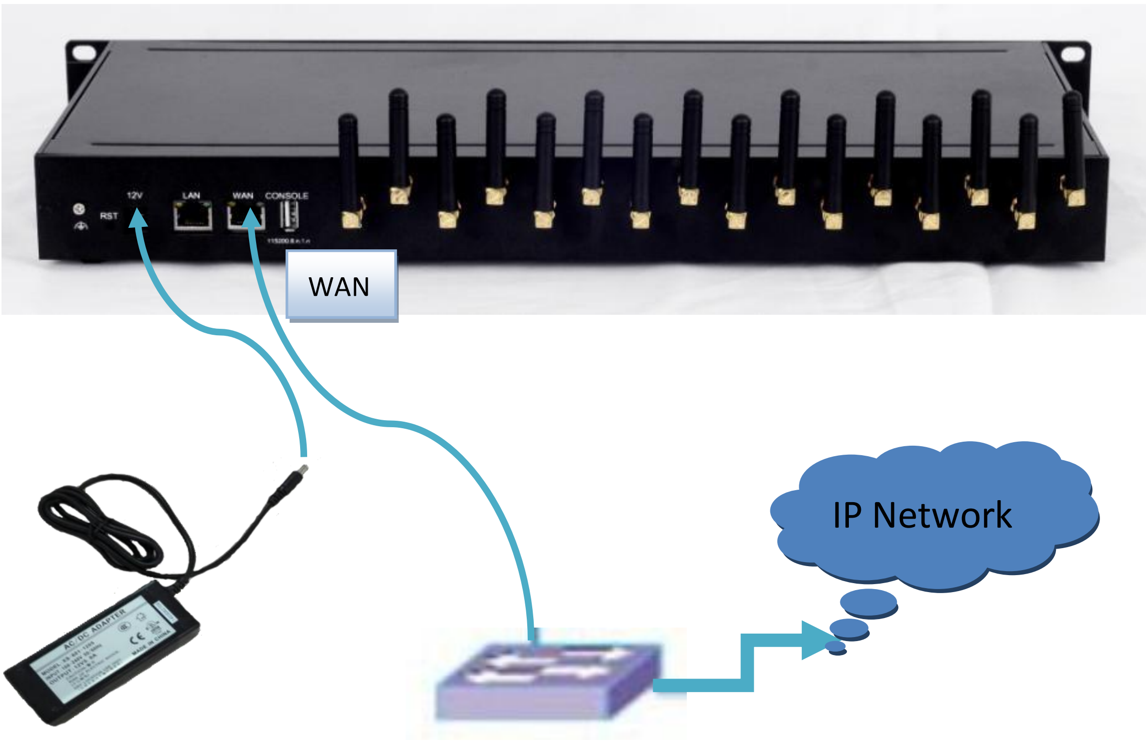
- Multi-language Interface
- USB Serial COM
- Configuration Backup and Restore
- Support HTTP/TFTP Upgrade
- Call statistics: ASR,ACD,PDD
- WEB Remote Management System

❖3 Equipment Installation

This chapter describes how to install a new GoIP Gateway to a physical network environment, how to initialize it and start it in a proper way.

3.1 Network Setup

Network is a prerequisite to install GoIP Gateway. The following figure shows the topology of Network with a VoIP Gateway connected.



Switch/Router

3.2 Equipment IP Address

The default IP of GoIP Gateway WAN port is 192.168.1.10, while the default LAN port IP is 10.10.10.1(The Version not up the LAN port).

3.3 Equipment Connection

Follow the steps below to install the GoIP Gateway to Network.

1) Fix the antenna to the GoIP Gateway. (Optional)

2) Insert SIM card(s) to slots.
3) Connect an Ethernet Cable to the WAN port of GoIP Gateway. The other end of the Ethernet Cable should be connected to Network port of route or switch.
4) Connect an Ethernet Cable to the Network port of GoIP Gateway. The other end of the Ethernet Cable should be connected to PC or other network device. (Optional)
5) Plug in the GoIP Gateway.

3.4 LED Indicators

There are a set of LED lights in the front of GoIP Gateway. Lights will be on or glittering when the GoIP Gateway is power on and running. The following table describes various meanings of status corresponding to LED lights in different display color.

Power	It indicates whether the system is running or not.
Calling	It indicates which module is calling.
Chosen card	It indicates was chosen.

❖4 Web Settings

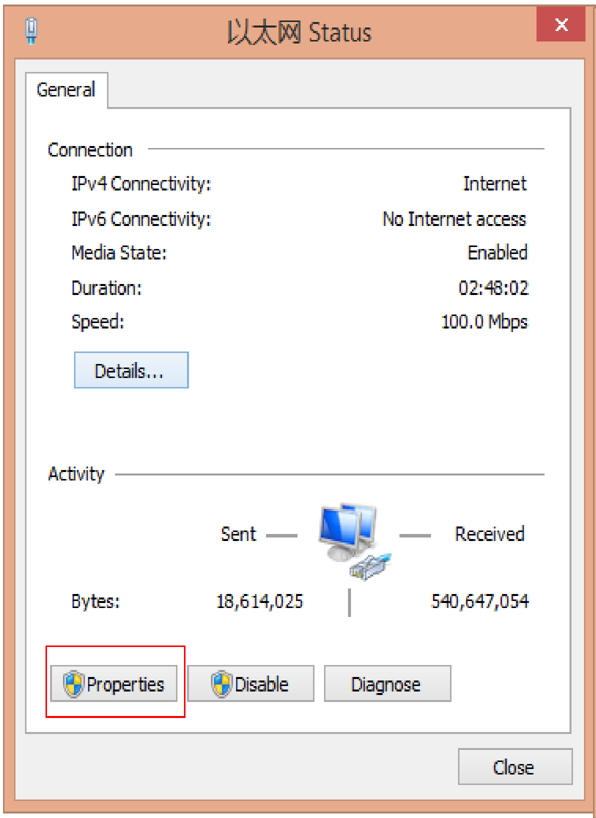
This chapter describes how to set up GoIP Gateway through Web Page. There is a built-in web server which can be accessed at URL: http://GATEWAY_IP/, while GATEWAY_IP is the WAN IP address of the GoIP Gateway, such as 192.168.1.10.

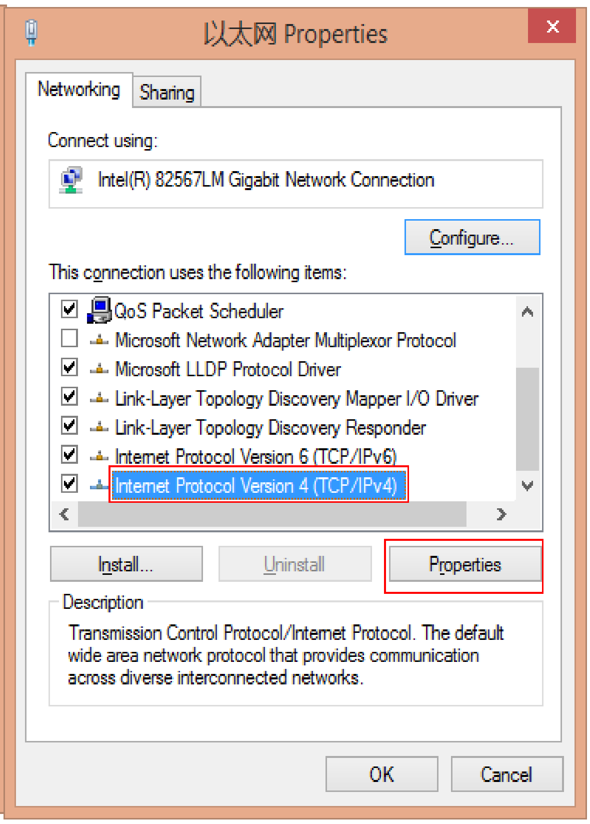
As an example, the following introduction will base on the GoIP Gateway with WAN IP 192.168.1.10.

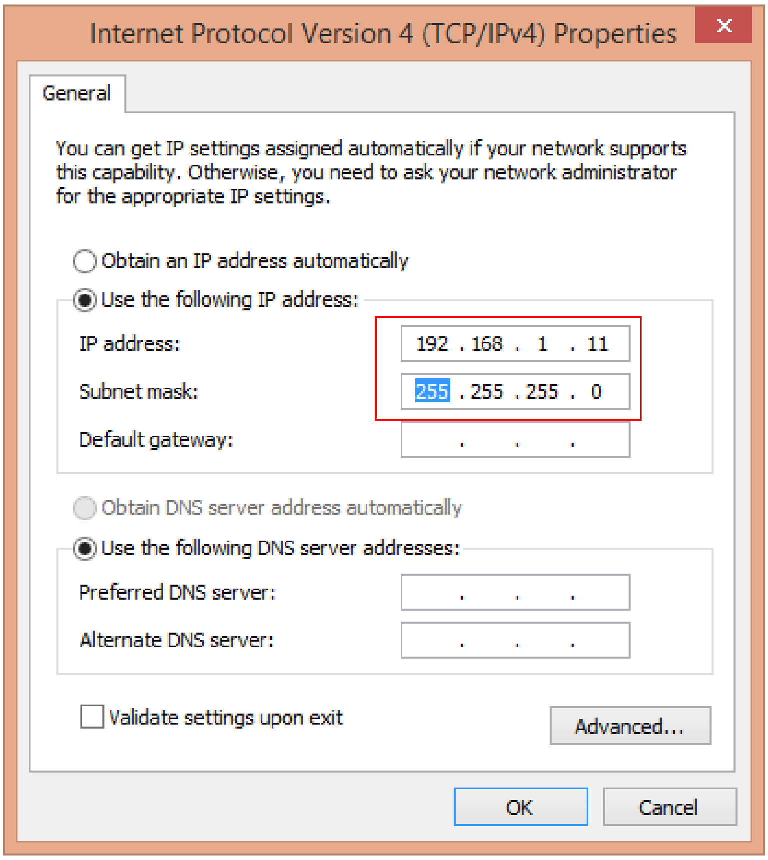
4.1 Login

First, connect a computer to the same LAN with GoIP, add the GoIP IP segment in the computer.

If your computer's IP address is not 192.168.1. XXX, how to change:

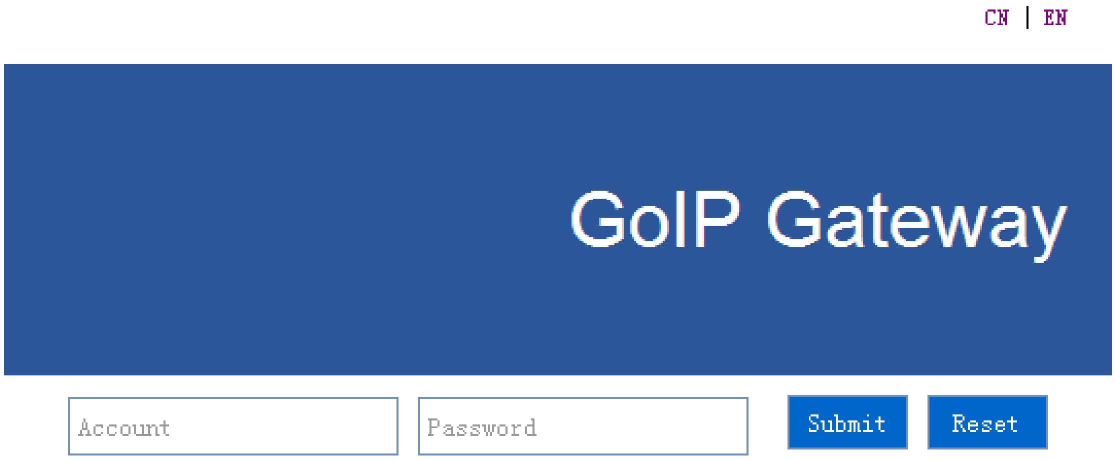






Save it

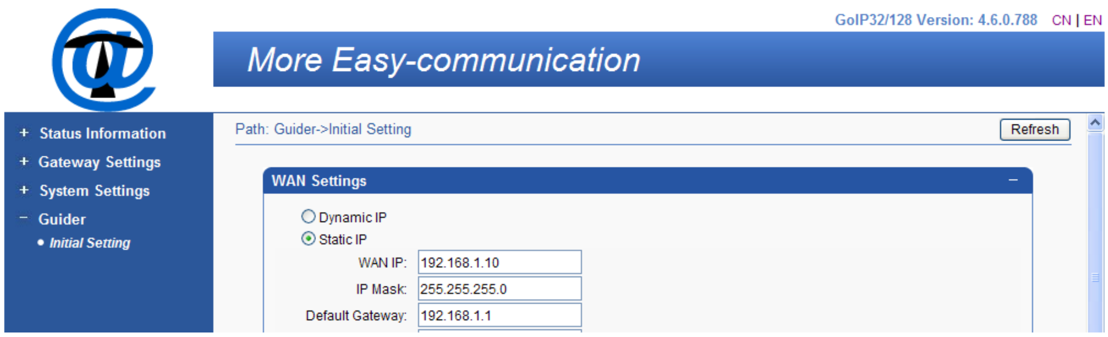
Open web browser and access URL http://192.168.1.10. The default login page will be displayed as following.



The default login account and password are:

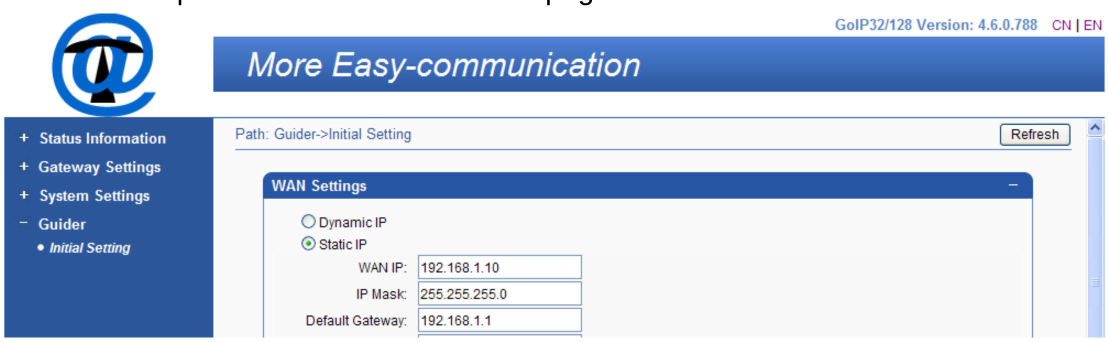
Account	root
Password	root

It is recommended to use IE or FireFox to access the web pages. After successfully logged in, the main page to set Gateway is as following:



❖5 Guider

Guider will be described in this paragraph. The most frequently modified parameters and most of the individual parameters are listed in this page.

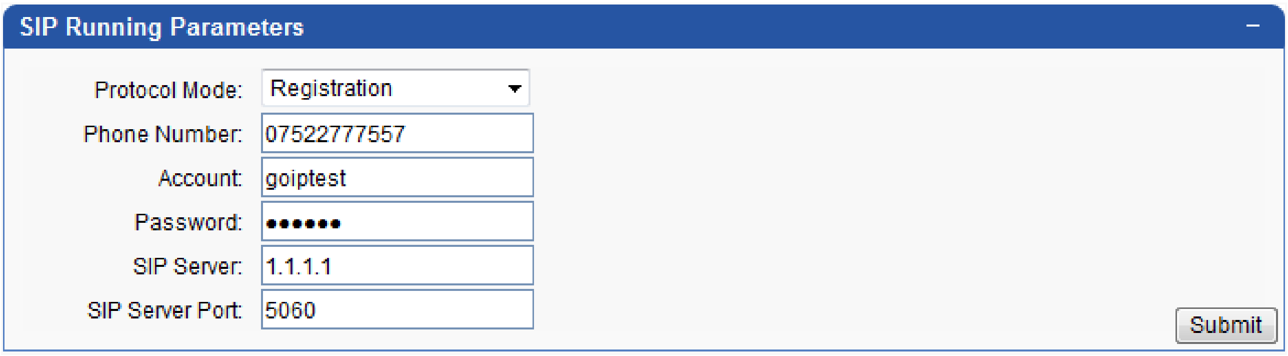


WAN port is to connect to switch or route then you can use PC that on the same LAN network to login. There's three types of wan connect.

Static IP: You can set WAN IP,IP Mask, Default Gateway and DNS Server of WAN port to connect the Internet.

Dynamic IP: You can get IP address ,IP Mask ,Default Gateway and DNS Server dynamically from your DHCP server.

PPPoE: In this mode,you can use GOIP device to dail-up network. You can set User Name and Password which you get from your ISP. And you also can set MTU and Service Name.



SIP Server Settings is for SIP communication with IP network. Fields are specified as following:

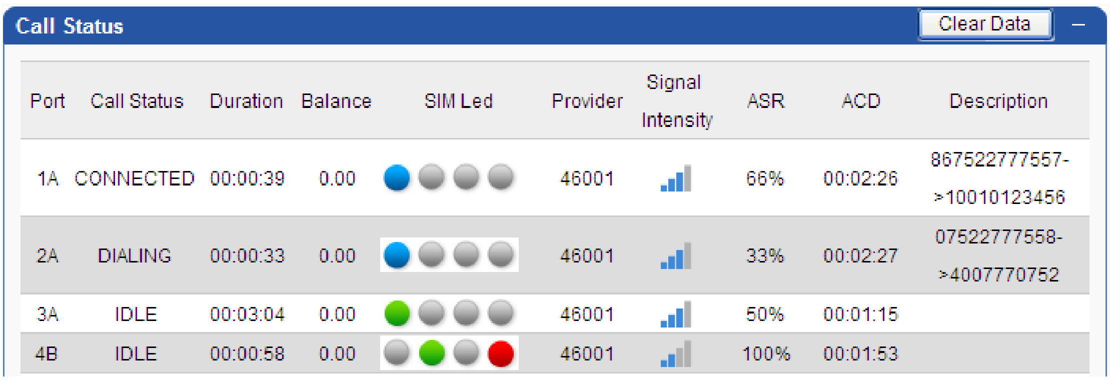
- Protocol Mode: Specify SIP client working mode. Option values are Registration/Point-to-Point. If set to Registration, SIP client will send registration messages to SIP server.
- SIP Server IP: Specify the IP of SIP server.
- SIP Server Port: Specify the port of SIP server.
- Phone Number: Specify the caller phone number for SIP client. It can also be regarded as the SIP port number which can be called.
- Account: Specify the SIP account for registration.
- Password: Specify the SIP password for registration.

Note: The settings of Phone Number, Account and Password are globally active. They will apply to all SIP ports settings in SIP Protocol page.

❖6 Status Information

6.1.1 Call Status

The screenshot below shows the live call status.



Port	Call Status	Duration	Balance	SIM Led	Provider	Signal Intensity	ASR	ACD	Description
1A	CONNECTED	00:00:39	0.00		46001		66%	00:02:26	867522777557->10010123456
2A	DIALING	00:00:33	0.00		46001		33%	00:02:27	07522777558->4007770752
3A	IDLE	00:03:04	0.00		46001		50%	00:01:15	
4B	IDLE	00:00:58	0.00		46001		100%	00:01:53	

The status columns are specified as following:

- Port No: The physical port sequence from 1 to 32.
- Call State: Specify the call status.
- Duration: Specify the call duration.
- Balance: Specify the current balance of the card in this port.
- SIM Led: shows the GoIP Gateway port LED status. Different LED
- Provider: The mobile provider that system detects.
- Signal Intensity: Specify the mobile signal intensity.
- ASR: Shows the port of ASR statistics.
- ACD: Shows the port of ACD statistics.
- Description: Specify the card dialing number status.
- Clear Data: Clear ASR and ACD data, restart statistics.

Note:

LED A/B/C/D displays in accordance with the lights on the front board of GoIP Gateway. Port 1 to 32 relate to the physical port of GoIP Gateway. The following table shows the relationship between LED color and port status.

Card Detected	Registering Card	Register OK	Calling
No Balance	Register Failed	Locked	Locked By Operator

6.1.2 Call Statistics

The screenshot below shows the call statistics information for analysis.

Call Statistics Clear Data

PortNo.	Calls	Alerted	Connected	Consecutive Fails	No Carriers	PDD	Tot CallDur
Total	593	236	162	0	1/498	00:00:06	11:00:31
1A	12	4	2	6	0/7	00:00:04	00:02:40
1B	49	17	13	2	0/42	00:00:05	00:36:05
1C	0	0	0	0	0/0		
1D	0	0	0	0	0/0		
2A	60	40	35	7	0/24	00:00:04	01:25:43
2B	0	0	0	0	0/0		
2C	0	0	0	0	0/0		
2D	0	0	0	0	0/0		
3A	18	10	6	2	0/18	00:00:04	00:35:21
3B	19	9	4	1	0/12	00:00:05	00:17:47

The status columns are specified as following:

- Port No: The physical port sequence from 1 to 32.
- Calls: Specify the total calls made out from this port since the last start up of system.
- Alerted: Specify the total number of responded alerting message for all the calls made.
- Connected: Specify the total number of answer from destination for all the calls made.
- Consecutive Fails: Specify the consecutive fail calls.

- No Carriers: Specify the no carrier.
- PDD: Specify the average duration to receive the response of alerting message.
- Tot CallDur: The port total calls duration used out from this port since the last start up of system.
- Clear Data: Clear all data, restart statistics.

A total summary is displayed at the bottom of the table.

6.2 Device Status(SIP and Module)

The screenshot below shows the SIP and Module status.

SIP Status

PortNo.	Registration Status	Module Status	IMEI
1	Ready	Yes	865328020699475
2		Yes	865328020694914
3		Yes	865328020695382
4		Yes	865328020694575
5		Yes	865328020731336
6		Yes	865328020729645
7		Yes	865328020725965
8		Yes	865328020730353
9		Yes	865328020693411
10		Yes	865328020700661

The status columns are specified as following:

- Port No: The physical port sequence from 1 to 32.
- Registration Status: Shows the port registration to SIP server infomation status.
- Module Status: Shows the port Module use status(Yes/No).
- IMEI: Specify the port current using of IMEI.

6.3 System Status

The screenshot below shows the system status. It includes WAN status, LAN status and others. The reported information can help you get the system status detail in a fast, simple way.

WAN Status

Connection Mode:	Static	Connection Status:	Connected
IP:	192.168.1.10	Default Gateway	192.168.1.1
DNS Server IP:	192.168.1.1	MAC Address:	00-30-f1-00-07-d3

LAN Status

IP:	10.10.10.1	IP Mask:	255.255.255.0
DHCP Server Status:	Enabled		

Other Status

Current Time:	2014-10-05 18:37:09 UTC+8	Running Time:	41 Hr 2 Min 14 Sec
Hardware Version:	1.0.0.0.1	Firmware Version:	0.4.2
Software Version:	532-460-788-041-100-070	Released Time:	Sep 23 2014 15:23:48 r1443

6.4 InterCall Statistics

Inter-Calling Statistics

Port	State	Duration	Inc. Calls	Out. Calls	Rcvd SMS	Sent SMS	Descriptions
1B	IDLE		19	18	0	0	
2B	IDLE		7	9	0	0	
3A	WAIT_CALLSTOP	02:02:18	10	8	0	0	

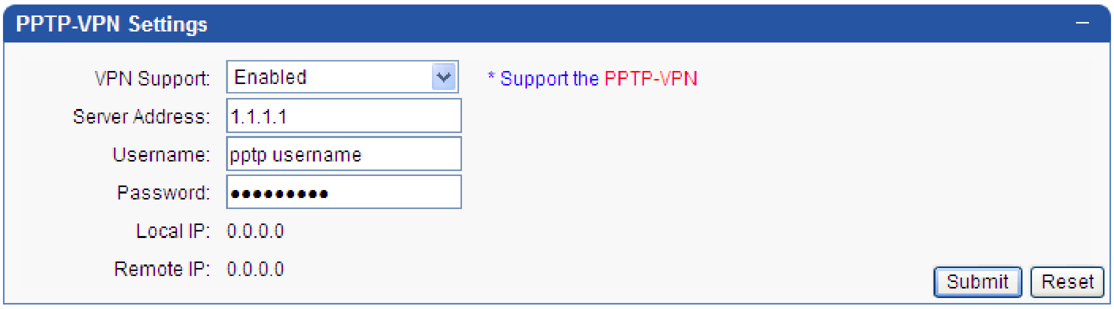
The status columns are specified as following:

- Port No: The physical port sequence from 1 to 32.
- State: Specify the call status.
- Duration: Specify the call duration.
- Inc calls: Specify the total incoming calls since the last start up of inter calling system.
- Out Calls: Specify the total outgoing calls since the last start up of the inter calling system.
- Rcvd SMS: Specify the total received SMSs since the last start up of the system.
- Sent SMS: Specify the total sent SMSs since the last start up of the system.
- Descriptions: Specify the card status.

❖7 Gateway Settings

7.1 Network Settings

The screenshot below shows the operation mode to set VPN settings, and the protocol of vpn is pptp.



Fields are specified as following:

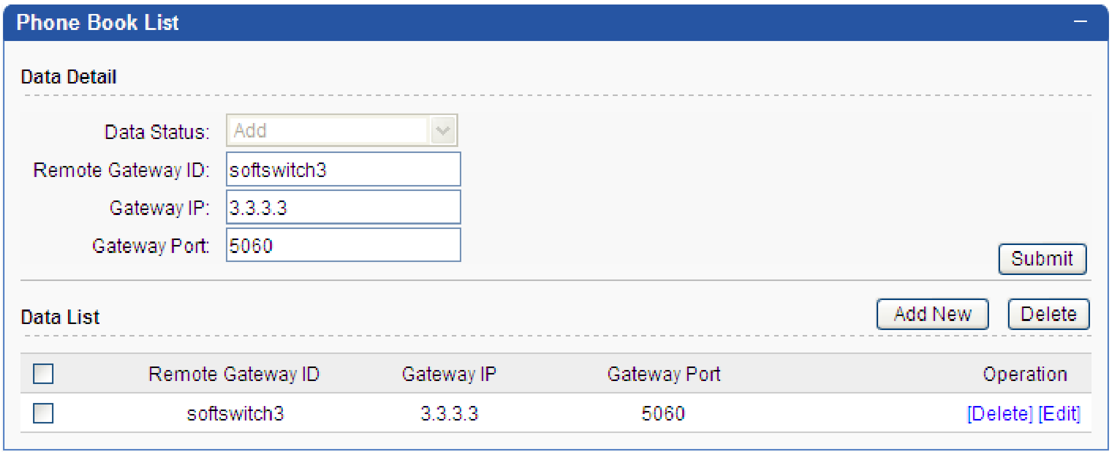
- VPN Support: Whether support vpn or not.
- Server Address: Specify the vpn server address.
- Usename: Specify the username of vpn.
- Password: Specify the password of vpn.
- Local IP: The VPN client ip.
- Remote IP: The VPN remote ip

The default port of web server is 80. The field Web Port is used to set another different port for web server. For example, if field Web Port is set to 8080 and wan IP is 192.168.1.10, the web pages then should be accessed through URL: http://192.168.1.10:8080 from this computer.

The field Telnet Port is used to change the default port of telnet service.

7.2 Phone Book(IP to IP calls)

The screenshot below shows the operation mode to set phone book. Phone book is a list contains the relationship between destination phone prefix and gateway information.



Add New

Click button Add New to expand the data input area to add new data. Fields are specified as following:

 Data status: Mark the status of current data record. Option values are Add/Edit. Value Add means the data is new while value Edit means the data is old.

- Remote Gateway ID: Specify the prefix of destination number for outbound call to IP. If a destination is best matched with any prefix in phone book, the destination call will be routed to the IP gateway specified by Gateway IP and Gateway Port.
- Gateway IP: The remote gateway IP.
- Gateway Port: The remote gateway port.

Click button Submit on the right to save the new data record.

Edit

All the phone book records are displayed in list. Two operations are provided on the right of each record. Click Edit to expand the current data record to Data Detail Area which is above the Data List.

Click button Submit on the right to save the old data record.

Delete

Click Delete on the right of each record to delete the current record. A message box will be popped for delete confirmation.

Another shortcut button is also provided on the top right of Data List to delete multiple selected records in batch. A message box will be popped for confirmation of batch delete.

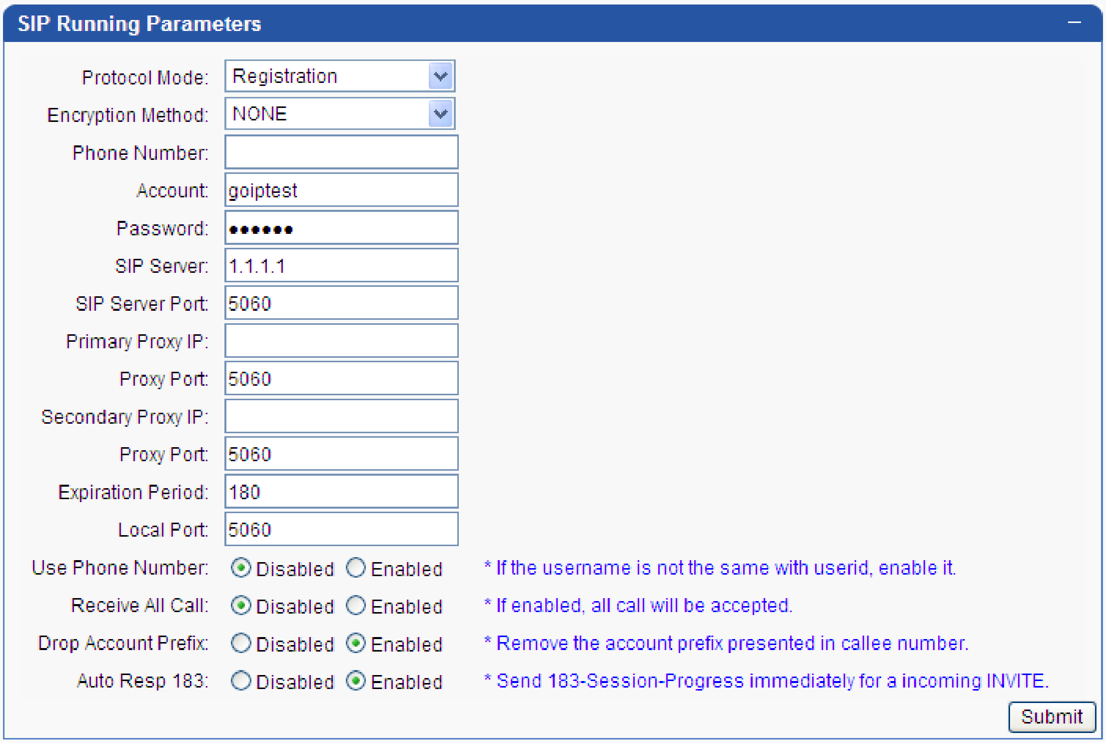
Note: The operations Add New/Edit/Delete/Batch Delete mentioned in the following paragraph are almost the same as phone book.

If you changed the data, take effect after reboot the gateway

7.3 SIP Setting

SIP Settings will be described in this paragraph. It mainly targets to set up parameters related to SIP server, SIP account and SIP password for SIP registration.

The screenshot below shows the operation mode to set SIP running parameters.

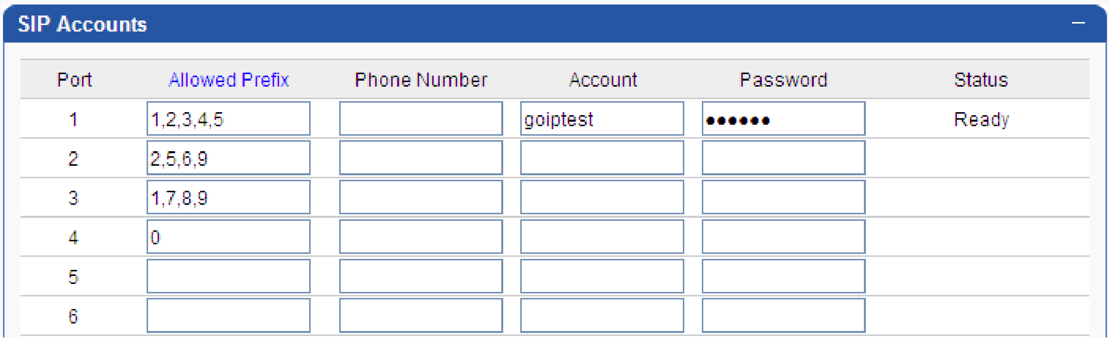


Fields are specified as following:

- Protocol Mode: Specify SIP client working mode. Option values are Registration/Point-to-Point/Multiple Port Support. If set to Registration, SIP client will send registration messages to SIP server. If set to Point-to-Point, o nly can let the input server IP traffic, if multiple IP, can increase in the phone book. If set Multiple Port Suppor, specify whether support to register to SIP server with different SIP Account, input in "SIP Accounts".
- Encryption Mode: Specify the encryption method for messages between SIP server and SIP client.
- Account: Specify the SIP account for registration.
- Password: Specify the SIP password for registration.
- SIP Server: Specify the domain or IP of the SIP Server.
- SIP Server Port: Specify the port of the SIP Server.
- Primary Proxy IP: Specify the primary proxy IP.
- Secondary Proxy IP: Specify the secondary proxy IP.
- Expiration Period: Specify the expiration period for registration.
- Local Port: Specify the local port used to register to SIP server.
- Use Phone Number: Specify whether enable phone number registration or not. If set to Enabled, the Phone Number and Account can be different when registering to SIP Server. If set to Disabled, the Phone Number must be the same as the Account when registering, otherwise, the registration will fail.

- Receive All Call: Specify whether enable to receive all calls or not.
- Drop Account Prefix: Specify whether enable drop account prefix. If set to enabled, it will remove the account prefix presented in callee number.
- Auto Resp 183: Specify whether enable auto resp 183 or not. If set to enabled,it will send 183-session-progress immediately for a incoming invite.

The screenshot below shows the operation mode to SIP Accounts:

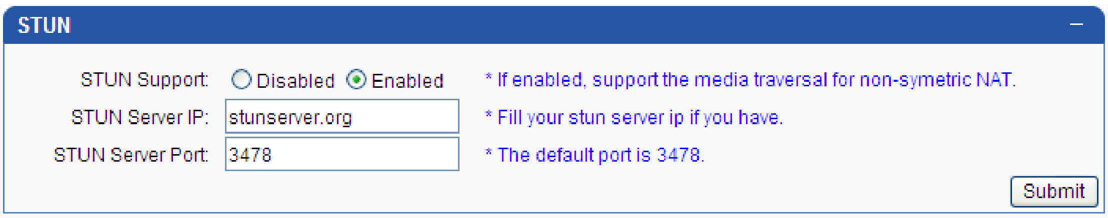


- Allowed Prefix: If empty, all prefixes can pass; If the input values, only allow input the prefix, if multiple prefix, separated by commas.

If a default Phone Number, Account and Password is already set in Guider->Initial Setting page, all the inputs here can be left empty and the default setting will apply to all the SIP accounts. For example, if an account “goiptest” is set with password “888888” in Guider->Initial Setting page and all the SIP accounts here are left empty, the combination of test and 888888 will be used for all SIP accounts to register to SIP Server. When the gateway register on the server successfully, the status will show ready.

There are two ways to overwrite the default SIP account setting in Guider Settings page.

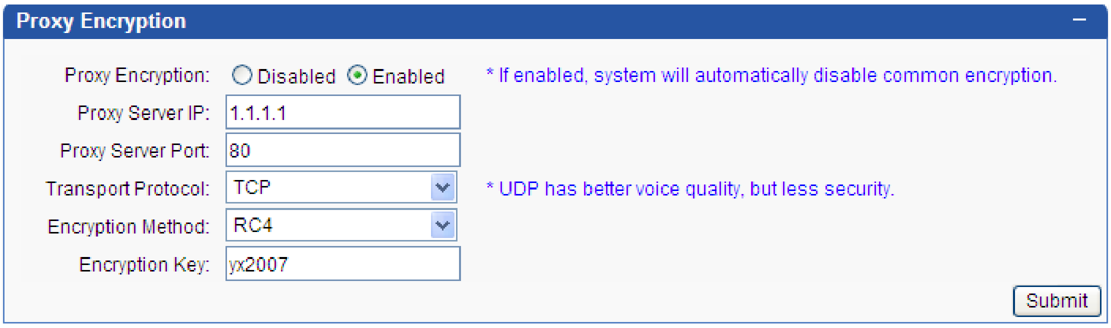
The screenshot below shows the operation mode to set STUN



STUN is a standardized set of methods and a network protocol to allow an end host to discover its public IP address if it is located behind a NAT. It is used to pemit NAT traversal for applications of real-time voice, video, messaging, and other interactive IP communications.

If you have the STUN server,enable STUN support,fill the server ip and port(default port is 3478), it will work.

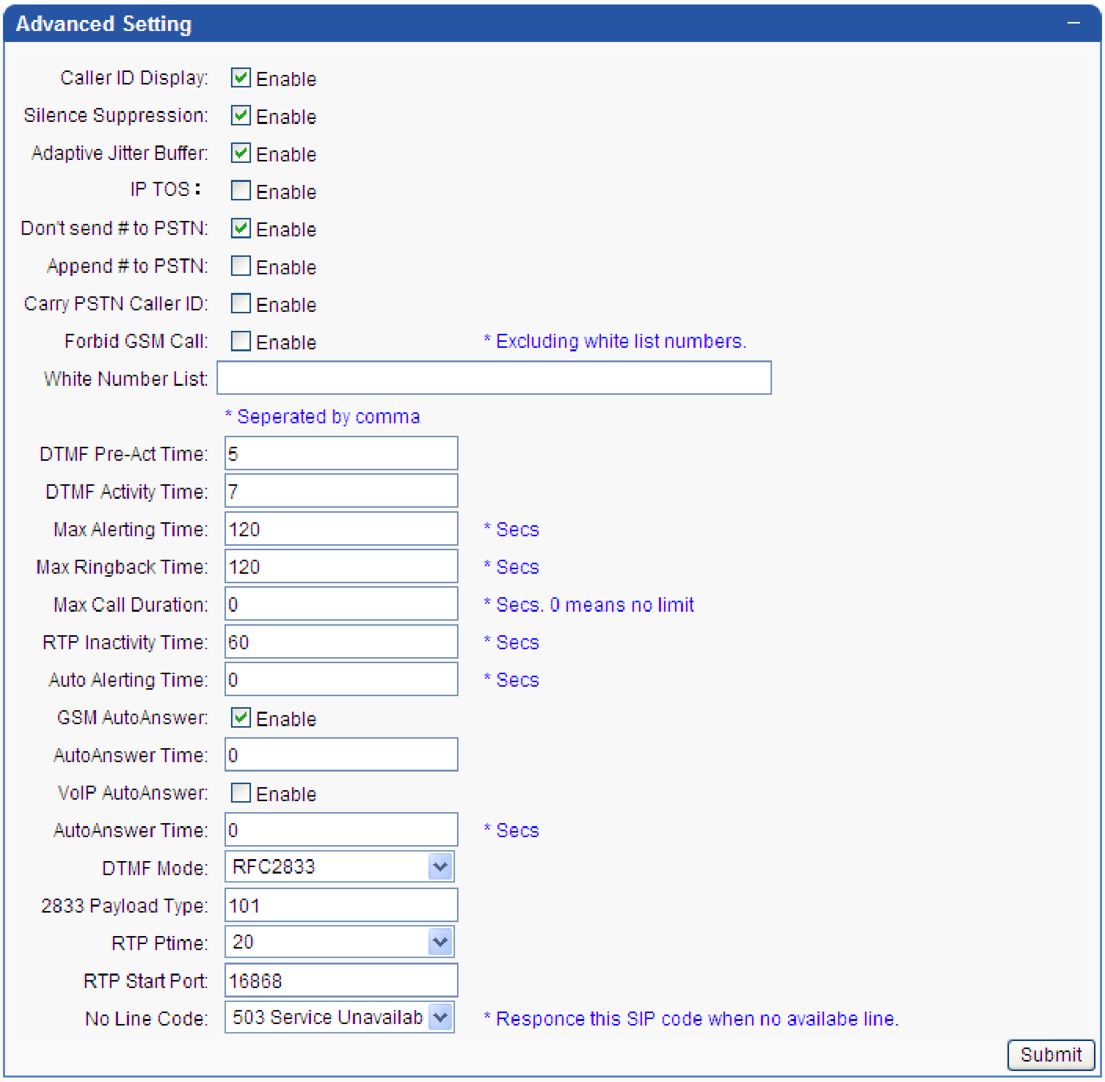
The screenshot below shows the operation mode to Proxy Encryption



Proxy Encryption is a good solution for protecting ip from blocking and save bandwidth about 50%. Fields are specified as following:

- Proxy Encryption: Specify whether enable it, if enabled, system will automatically disable common encryption.
- Proxy Server IP: Specify the proxy server ip.
- Proxy Server Port: Specify the port of the proxy server, the default proxy server port is 38080.
- Transport Protocol: Specify the transport protocol. Option values are UDP/TCP,UDP has better voice quality, TCP is more security.
- Encryption Method: Specify the encryption method, the default encryption method RC4,other two method are XOR and BASE64.
- Encryption key: Specify the encryption key.

The screenshot below shows the operation mode to set application feature：



The fields are specified as following:

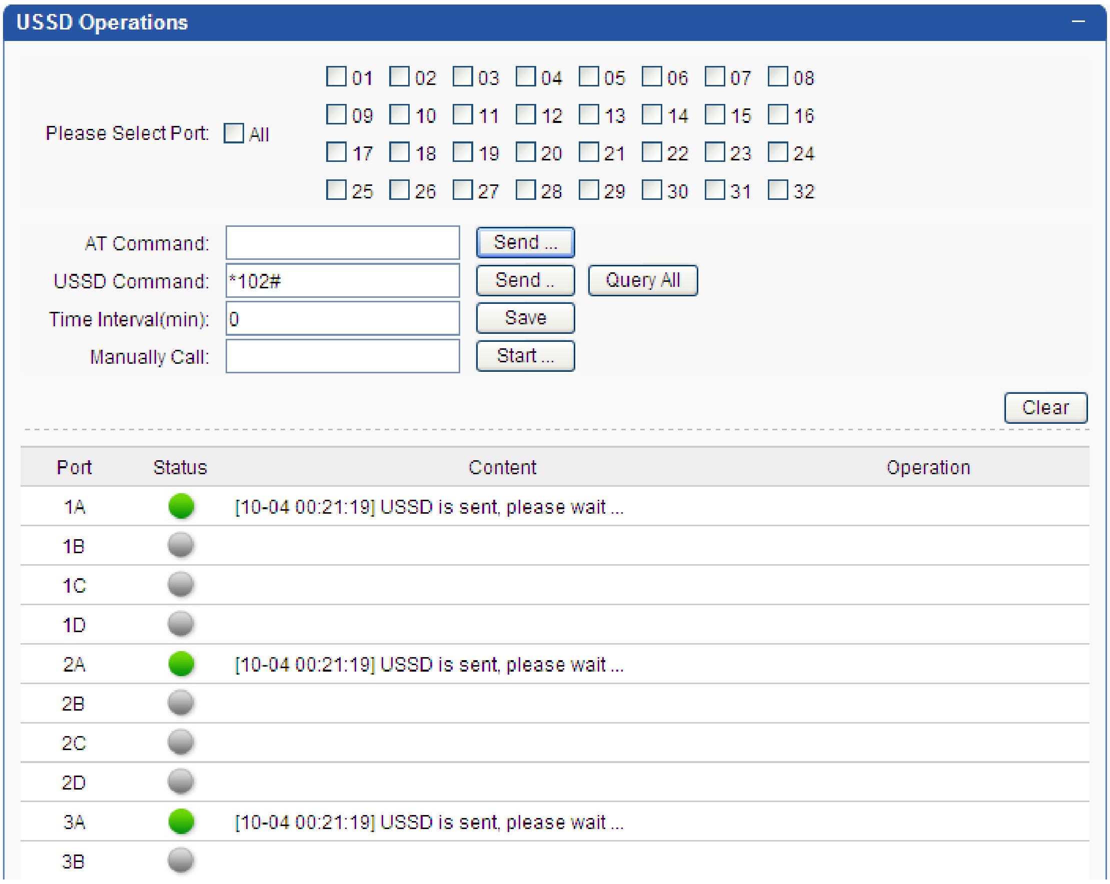
- Caller ID Display: Specify whether enable Caller ID Display or not.
- Silence Suppression: Specify whether enable Silence Suppression or not.
- Adaptive Jitter Buffer: Specify whether enable Jitter Buffer or not.
- IP TOS: Specify whether enable IP TOS or not.
- Don't send # to PSTN: Specify whether need to remove the last digit # to PSTN if the last digit of numbers is #. If set to Enable, the last # will be removed.
- Append # to PSTN: Specify whether need to send an extra # to PSTN after the normal digital numbers are sent.
- Carry PSTN Caller ID: Specify whether need to carry PSTN caller ID to system.
- Forbid GSM Call: Specify whether need to prevent the PSTN incoming call. If set to enable, any of the incoming calls whose callerid is not in the white list specified in White Number List will be prevented.
- White Number List: Specify a caller number list separated by comma. It is used in combination with Forbid GSM Call to let the specified caller pass through and continue the incoming call flow if Forbid GSM Call is set to Enable.
- DTMF Pre-Act Time: Specify the DTMF Pre-Act Time.
- DTMF Activity Time: Specify the DTMF activity time.
- Max Alerting Time: Specify the max alerting duration.

- Max Ringback Time: Specify the max ringback duration.
- Max Call Duration: Specify the max call duration. System will hang up the call automatically if the call duration reaches this value.
- RTP Inactivity Time: Specify the max duration of silence from gateway. System will hang up the call automatically if the silence duration reaches or exceeds this value.
- GSM Call AutoAnswer: Specify whether need to auto-answer the call which is from GSM. If set to Enable, AutoAnswer Time can be used to specify the delay to automatically answer the incoming GSM call.
- VoIP Call AutoAnswer: Specify whether need to auto-answer the call which is from IP network. If set to Enable, AutoAnswer Time can be used to specify the delay to automatically answer the incoming IP call.
- DTMF Mode: Specify the DTMF mode Option values are RFC2833/Inband/SIP INFO.
- RFC2833 Payload Type: Specify the RFC2833 DTMF Payload Type. Only valid when DTMF Mode is set to RFC2833. The default value is 101.
- RTP Ptime: Specify the interval of RTP packages.
- RTP Start Port: Specify the RTP start port.

7.4 Port Setting

7.4.1 USSD Operations

The screenshot below shows the operation mode of USSD operation to GoIP Gateway.

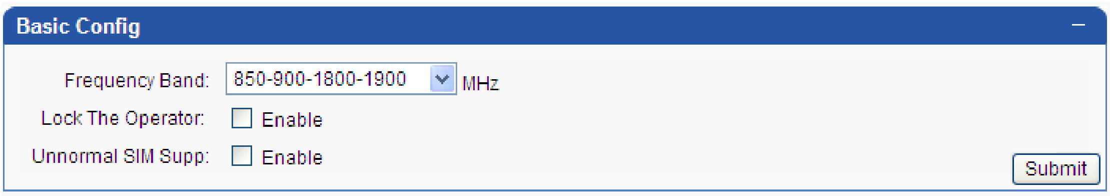


Fields are specified as following:

- Select port: Choose some or all port to execute AT or USSD command.
- At Command: You can enter AT command then execute to the port which you select.
- USSD Command: Enter USSD query command.
- Time Interval: Query balance regularly.
- Manually Call: Manually input a number, let the sim card mark a call

7.4.2 Base Station

The screenshot below shows the operation mode to set Basic Settings of Port settings



It’s for choosing the frequency band and whether lock the operator, generally, we don’t enable the lock the operator. When the gateway can’t detect the sims, we need to enable Unnormal Sim Support.

7.4.3 Hardware properties

The screenshot below shows the operation mode to set Hardware properties.

Hardware Properties

PortNo.	Type	Disable	Bind SIM Card	InputVol	OutputVol	Balance	Operations
1	GSM	☐	☑A ☑B ☑C ☑D	0	12	0.00	Restart More>>
2	GSM	☐	☑A ☑B ☑C ☑D	0	12	0.00	Restart More>>
3	GSM	☐	☑A ☑B ☑C ☑D	0	12	0.00	Restart More>>
4	GSM	☐	☑A ☑B ☑C ☑D	0	12	0.00	Restart More>>
5	GSM	☐	☑A ☑B ☑C ☑D	0	12	0.00	Restart More>>
6	GSM	☐	☑A ☑B ☑C ☑D	0	12	0.00	Restart More>>
7	GSM	☐	☑A ☑B ☑C ☑D	0	12	0.00	Restart More>>
8	GSM	☐	☑A ☑B ☑C ☑D	0	12	0.00	Restart More>>
9	GSM	☐	☑A ☑B ☑C ☑D	0	12	0.00	Restart More>>
10	GSM	☐	☑A ☑B ☑C ☑D	0	12	0.00	Restart More>>

Hotline: Unconditional Forward:

Busy Forward: NoAnswer Forward:

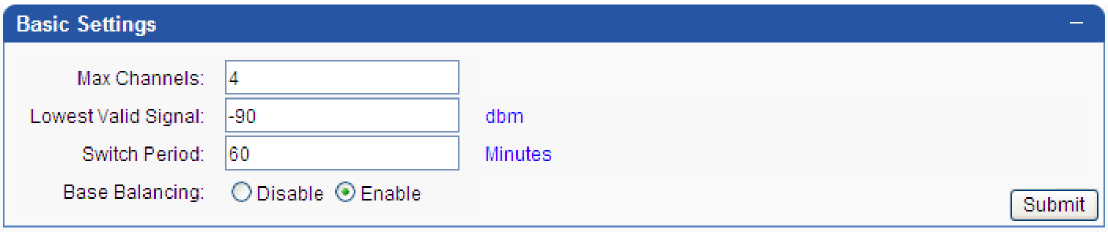
The columns are specified as following:

- Port No: The GoIP Gateway mobile port. Each port contains one or four card slots. Port No starts from 1 to 32.
- Type: Values are GSM/CDMA/WCDMA. (According to your module.)
- Disable: Specify whether enable or disable this port.
- Bind SIM Card: The SIM card that not bind will be locked by gateway.
- Input Volume: Specify the input voice volume of this port.
- Output Volume: Specify the output voice volume of this port.
- Balance: Shows the current balance of sim card
- Operations:
  - Restart: Restart the module
  - More:
    - Hot-line: Specify a sip phone on the softswitch to pick up the incoming calls
    - UnconditionalForward/NoAnserForwardNumber/BusyForward: These parameters are designed to be used with a third party system.

7.5 Basic Station

7.5.1 Basic Settings

The screenshot below shows the operation mode to set globally for base settings.

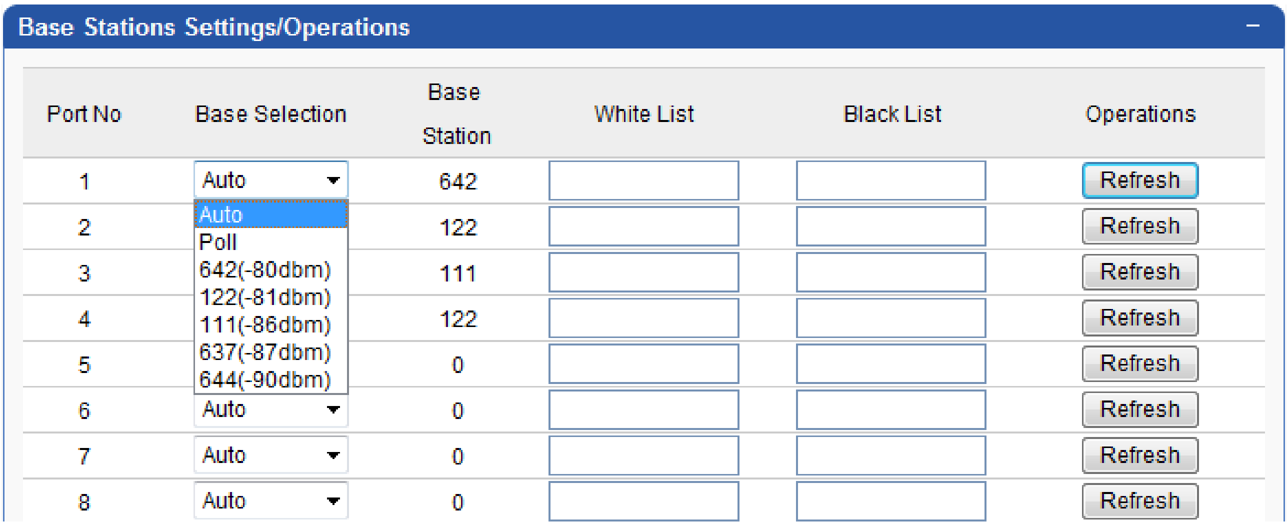


Fields are specified as following:

- Max Channels: Specify the max base stations
- Lowest Valid Signal: Specify the lowest valid signal
- Switch period: Specify the period of switching base station
- Base Balancing: Specify whether enable base balancing, we suggest disable it.

7.5.2 Base stations Settings/Operations

The screenshot below shows the operation mode to set base stations settings and operations



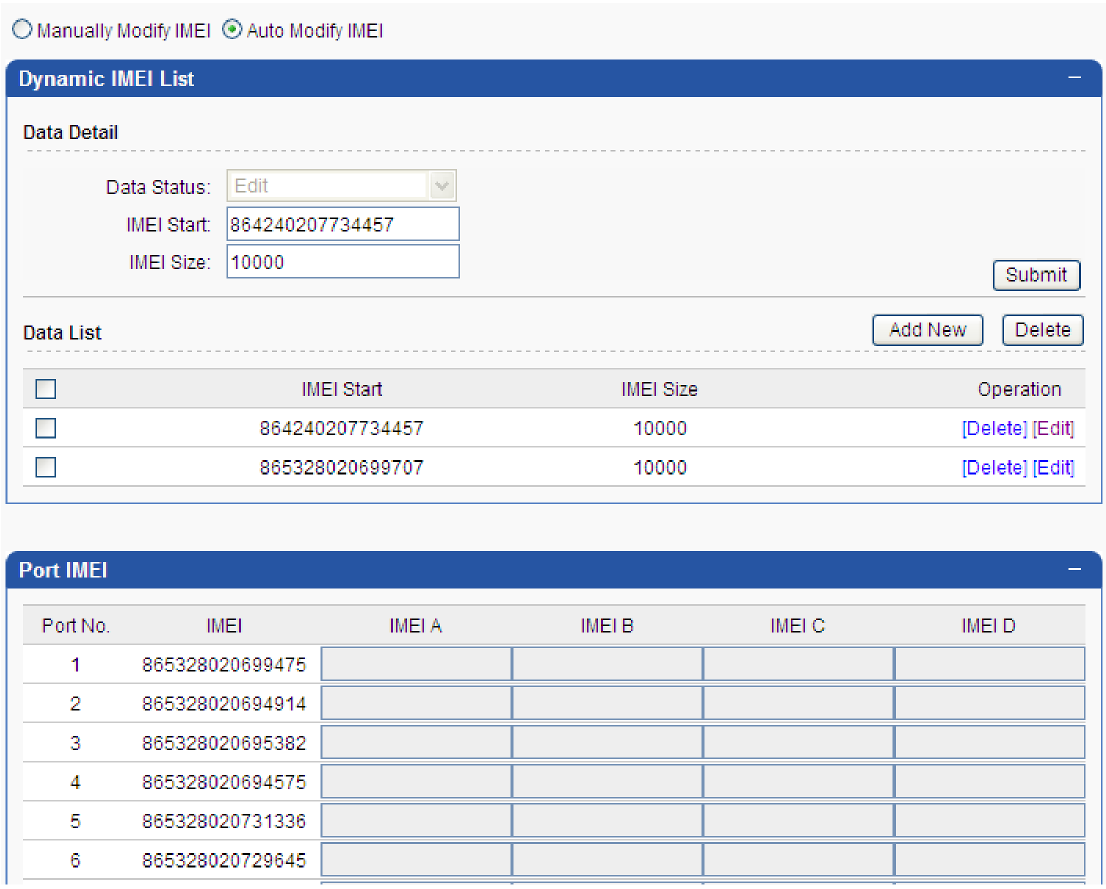
- Port No: The GoIP Gateway mobile port. Each port contains one or four card slots. Port No. starts from 1 to 32.
- Base Selection: the default base selection mode is auto which makes the devices to choose mobile base automatically, if you want to switch base station periodically, pleasure select Poll, and you can set the switch period.
- Base Station: The current using of base station.
- White List: Specify the white list of base station, can be multiple, use a comma.
- Black List: Specify the black list of the base station, can be multiple, use a comma.
- Operations: When click the refresh button will change a new base station.

7.6 IMEI Setting

The screenshot below shows the operation mode to set IMEI for each card inserted in GoIP Gateway port.

The specified IMEI, instead of the default IMEI of the card, will be used for the corresponding card to communicate with mobile base.

The screenshot below shows the operation mode to set Dynamic IMEI for each card of the designated port. If a card on a port is assigned with a group of IMEIs, it will randomly use any of the IMEI in group to communicate with mobile base.



- Manually Modify IMEI: Manually modify IMEI when the module corresponding to using the sim card
- Auto Modify IMEI: When rotating the sim card or restart the device, automatically change a IMEI (need to set up have IMEI data)

Add New

Click button Add New to expand the data input area to add new data. Fields are specified as following:

- Data Status: Mark the status of current data record. Option values are Add/Edit. Value Add means the data is new while value Edit means the data is old.
- IMEI Start: Specify an initial IMEI value for the IMEI group.
- IMEI Size: Specify the size of the IMEI group.

Click button Submit on the right to save the new data record.

Edit

All the records are displayed in list. Two operations are provided on the right of each record. Click Edit to expand the current data record to Data Detail Area which is above the Data List.
Click button Submit on the right to save the old data record.

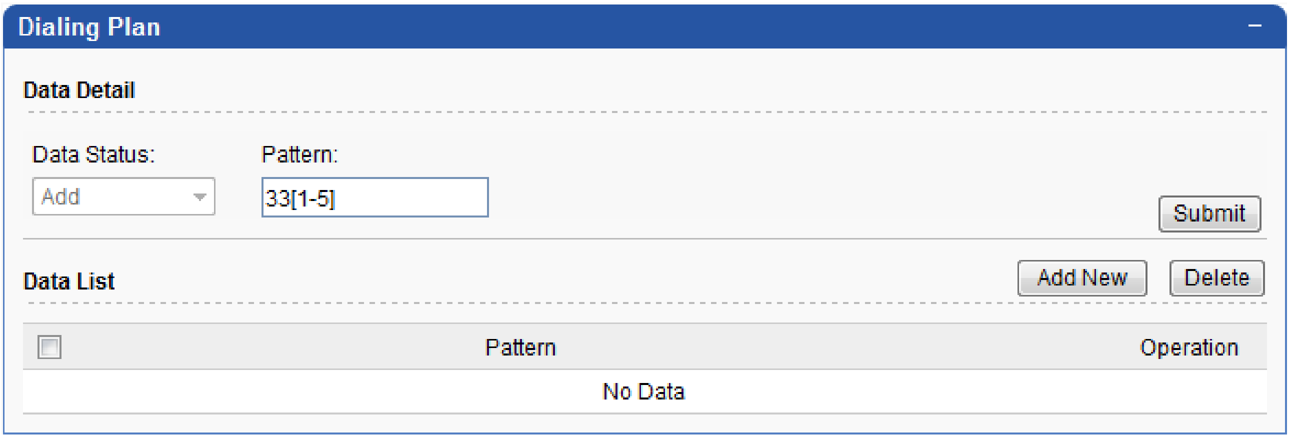
Delete

Click Delete on the right of each record to delete the current record. A message box will be popped for delete confirmation.

7.7 Rules Setting

7.7.1 Dialing Plan

The screenshot below shows the operation mode to set Dial Plan.



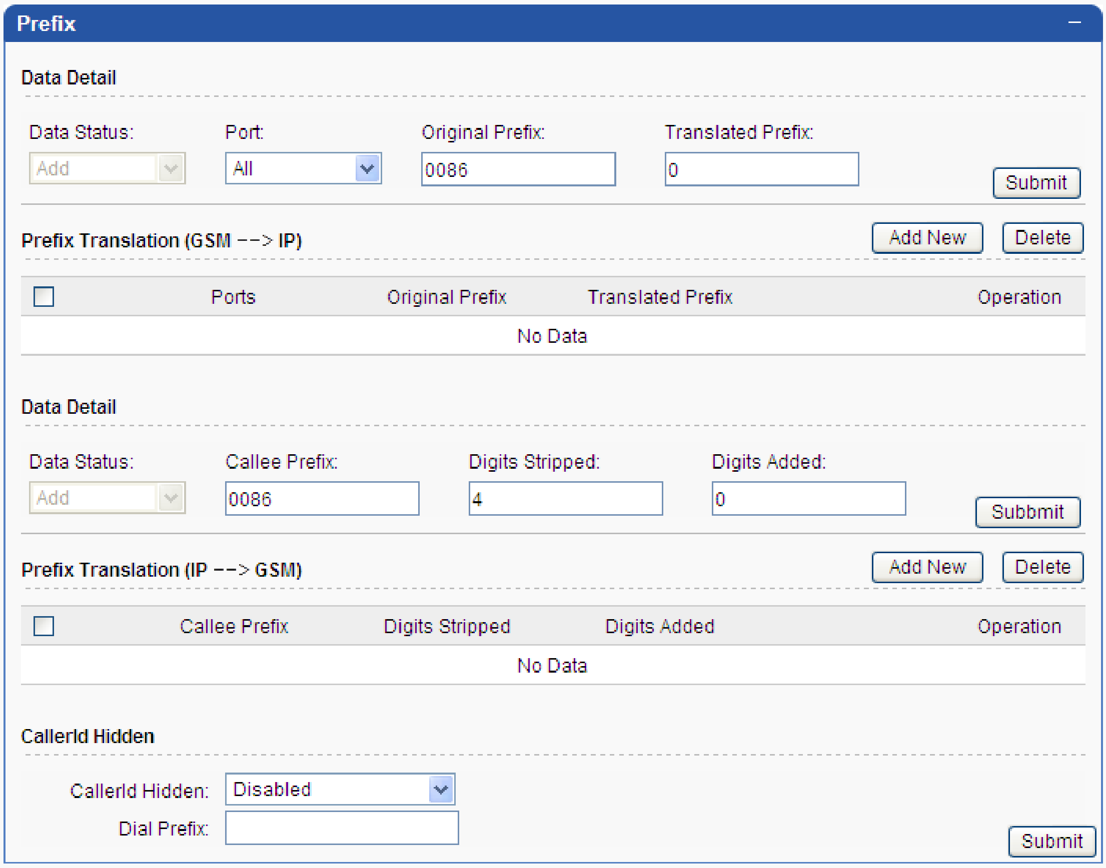
Dial Plan is a string which specifies digits and length of the digits of the dialed number. Generally, the pound sign # is used as the termination of number input for dialing. However, if patterns are specified and system detects the dialed number matches any of the patterns, it will stop collecting input and send out the collected number to dial even though no pound sign is encountered.

The dial plan string is a normal regular expression, for example:

The plan 33[1-5] means the dialed number starts with 33 and the last digit is any of 1/2/3/4/5. So any of the input 331, 332, 333, 334 or 335 is acceptable.

7.7.2 Prefix

The screenshot below shows the operation mode to set manipulation for dial prefix.



Fields are specified as following:

- Prefix: The original prefix in phone number.
- Manipulated Prefix: Specify the digits with which the value specified by Prefix will be substituted.

GSM-->IP

Take the value in screenshot as an example, the prefix 0086 in dialed number will be substituted with 0. That's to say, if 00867522777557 is input to dial, the final number dialed out is 07522777557.

Note: the manipulation is executed after pattern is matched.

IP-->GSM

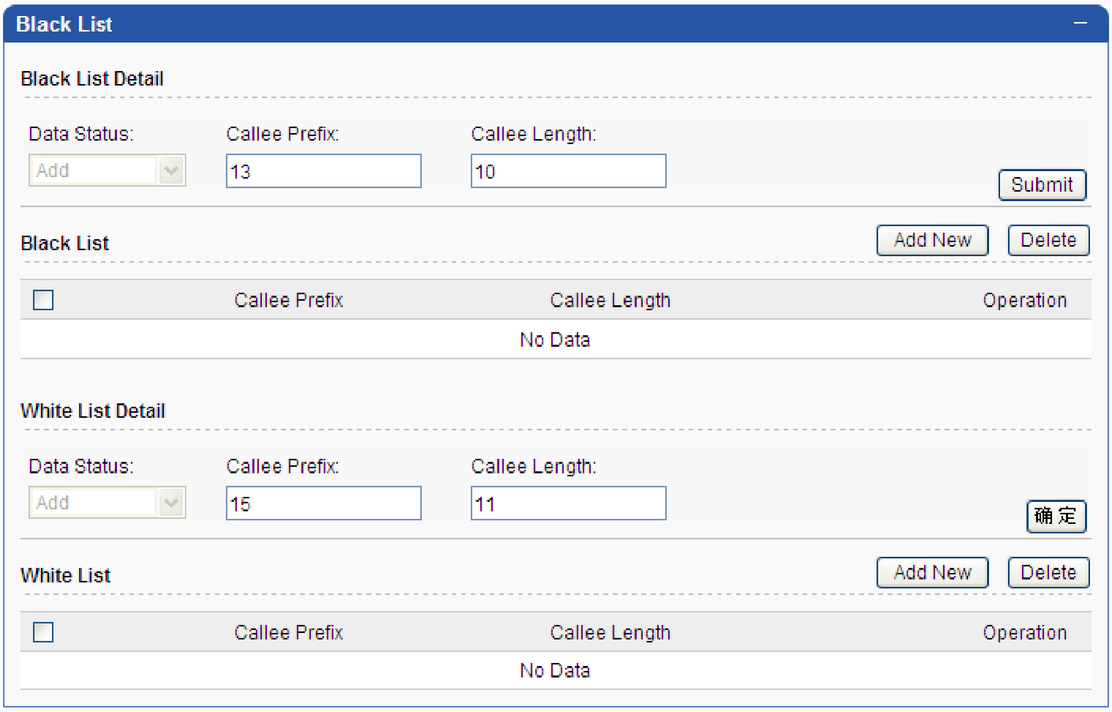
- Callee Prefix: Specify the callee prefix.
- Digits Stripped: Specify the digits which are stripped.
- Digits Added: Specify the digits which are added.

Take the value in screenshot as an example, the callee ID is 0086xxxxx, we specify 0086 as the callee prefix,strip 4 digits and add 0,when dial 0086123456789,the gateway will strip 0086 and add 0 call 0123456789.

CallerId Hidden

- CallerId Hidden:Enable or disable caller ID hidden.Your carrier must support that function.
- Dial prefix:Use the dial prefix you set when enable caller ID hidden.

7.7.3 Calls black and White



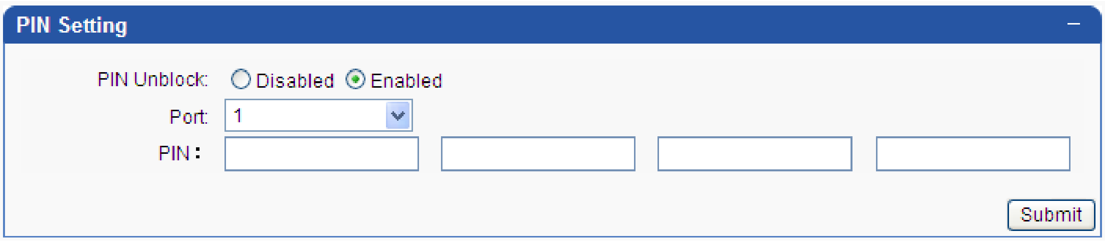
Fields are specified as following:

- Callee Prefix: Specify the callee prefix of black list.
- Callee Length: Specify the callee length of black list.

7.8 Mobile Setting

7.8.1 PIN Setting

The screenshot below shows the operation mode to set globally for PIN settings

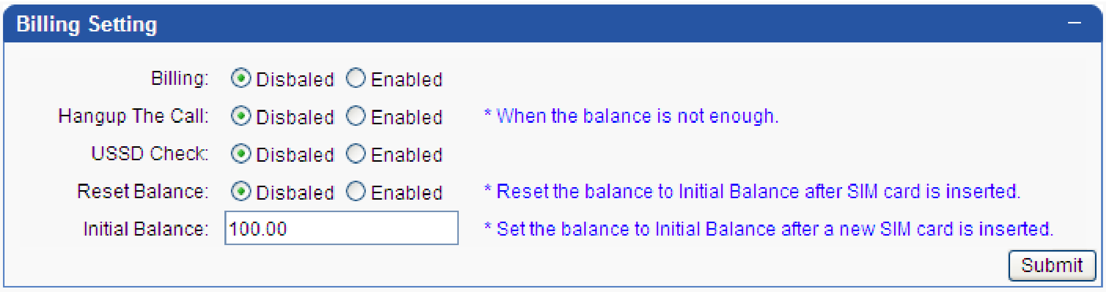


Fields are specified as following:

- PIN Unblock: Specify whether enable the pin unblock.
- Port: starts from 1 to 32
- PIN: Specify the PIN for card A/B/C/D of the port

7.8.2 Billing Setting

The screenshot below shows the operation mode to set GoIP billing. A smart billing server for mobile port is embedded in GoIP Gateway.

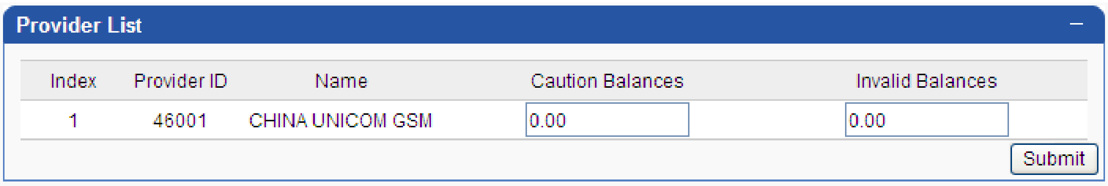


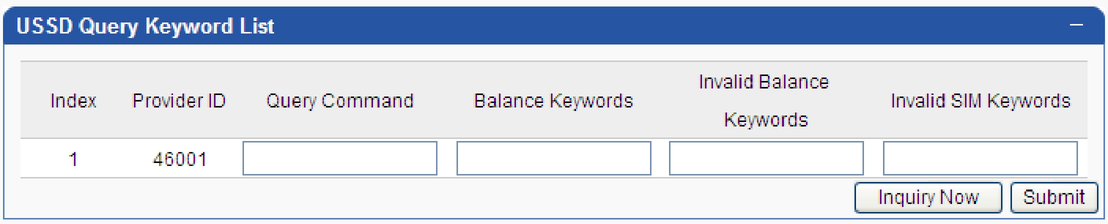
Fields are specified as following:

- Billing: Specify whether enable GoIP billing or not. If set to Enabled, system will bill the outbound calls for the port which has been assigned with billing tariffs.
- Hangup The Call: Specify whether enable hangup the call when balance is ont enough. When select enable, the call will be hang up immediately when run out of balance. But when you select disable the call will not be hangup.
- USSD Check: Specify whether enable to get balance through USSD check or not. This field takes effect only when GoIP Billing is set to Enabled.
- Reset Balance: Specify whether need to reset the balance to intial balance after sim card is inserted.
- Initial Balance: Special the initial balance for a new card.

The screenshot below shows the operation mode to set Caution Balances, Invalid Balances and USSD Keyword List. These settings are used for both getting balance through USSD and billing GoIP calls. The provider ID is detected by GoIP Gateway automatically. For a new Gateway without any card inserted, there may be no records in the two lists.

7.8.3 Provider and USSD Query Keyword



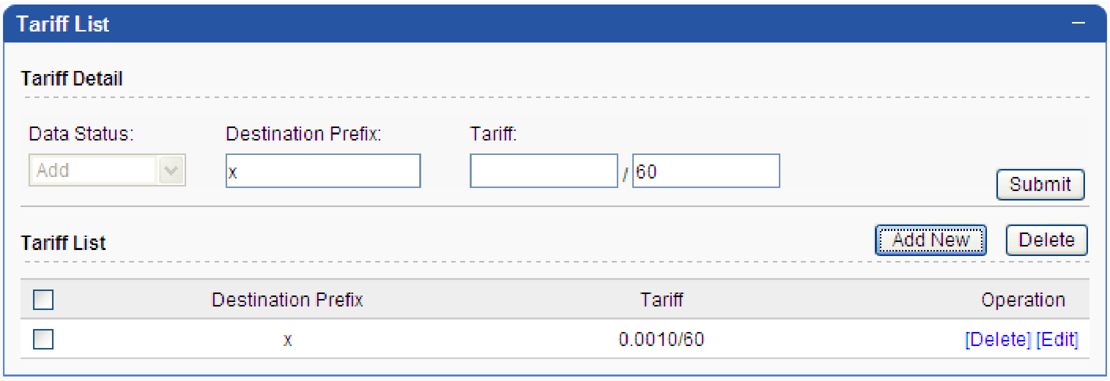


Fields are specified as following:

- Name: Specify the provider name.
- Caution Balances: Specify four balances separated by semi-colon. Each balance corresponds to one Keyword from left International Keys to right Other Keys in Keyword List table. Each caution balance is a threshold for system to get card balance through USSD if current balance is less than this threshold. However, the final card balance is based on only one type which is specified by field Billing Type in the part of Basic Settings.
- Invalid Balances: like Caution Balances, it specifies the threshold for system to disable the card if current balance is less than this threshold.
- Query Command: Specify the query command
- Balance Keywords: Specify keywords for system to analyze the balance data after sending USSD command to carrier mobile network.
- Invalid Balance Keywords: Specify the invalid balance keywords
- Invalid SIM keywords: Specify the invalid sim keywords.

7.8.4 Tariff List

The screenshot below shows the operation mode to set billing tariff.



Add New

Click button Add New to expand the data input area to add new data. Fields are specified as following:

- Data Status: Mark the status of current data record. Option values are Add/Edit. Value Add means the data is new while value Edit means the data is old.
- Destination Prefix: Specify the destination prefix used to bill call. If this prefix is best matched with a destination of an outgoing call from the port(s), the corresponding tariff will be chosen to bill the call. The prefix can be a regular expression. For example, [2-8] matches any phone number which starts with digit 2 to 8. And [0-9] matches all phone numbers.
- Tariff: Specify the tariff detail. In the picture,the tariff means 10 value will be deduct per 60 seconds.

Click button Submit on the right to save the new data record.

Edit

All the records are displayed in list. Two operations are provided on the right of each record. Click Edit to expand the current data record to Data Detail Area which is above the Data List.

Click button Submit on the right to save the old data record.

Delete

Click Delete on the right of each record to delete the current record. A message box will be popped for delete confirmation.

Another shortcut button is also provided on the top right of Data List to delete multiple selected records in batch. A message box will be popped for confirmation of batch delete.

7.9 SMS Setting

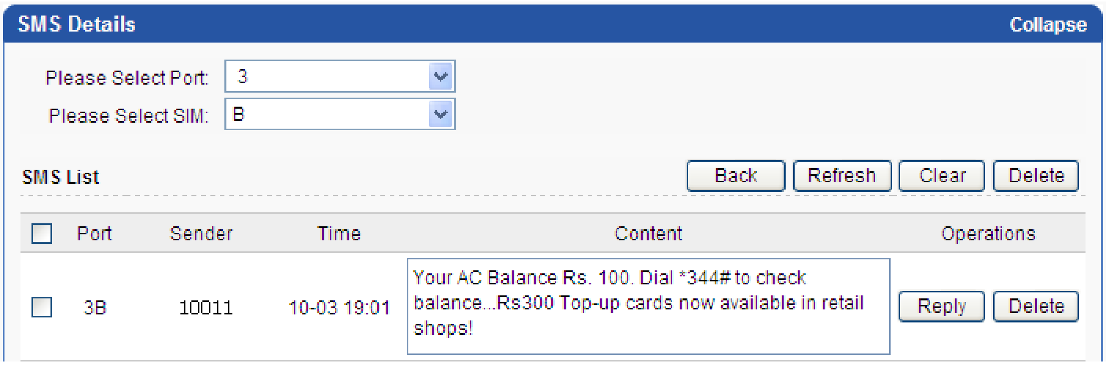
7.9.1 SMS Receive

The screenshot below shows the operation mode to receive sms.

Fields are specified as following:

- Port No: The GoIP Gateway mobile port. Each port contains one or four card slots. Port No starts from 1 to 32.
- Sender: Specify the sms sender.
- Time: Specify the sms receive time.
- Content: Specify the sms content.
- Operations: Click the Detail button to get more detail about the specify port.

The screenshot below shows the operation mode to get sms details.

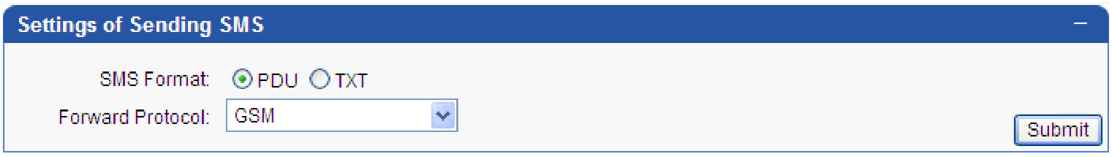


Fields are specified as following:

- Please Select Port: Specify the port.
- Please Select SIM: Specify the sim.
- Port: Specify the port.
- Sender: Specify the sms sender.
- Time: Specify the sms receive time.
- Content: Specify the sms content.
- Back: Back to the SMS content web page.
- Refresh: Refresh the web page.
- Clear: Clear the sms.
- Reply: Reply the sms
- Delete: Delete the corresponding sms.

7.9.2 SMS Send

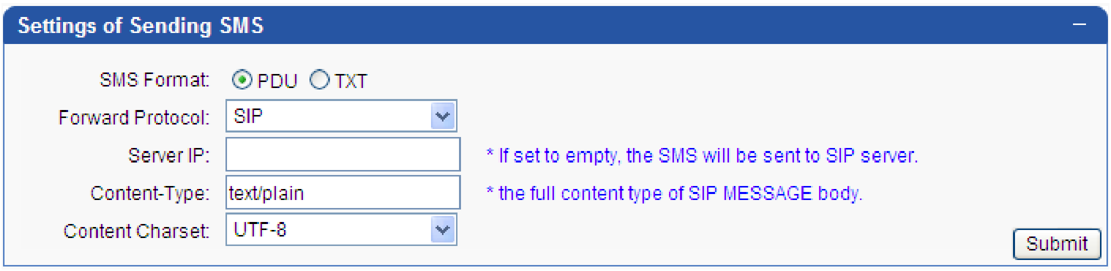
The screenshot below shows the operation mode to set basic settings.



Fields are specified as following:

- SMS Format: Specify the sms format.
- Forward Protocol: Specify forward protocol.

The screenshot below shows the operation mode to set sip protocol of forwarding sms.

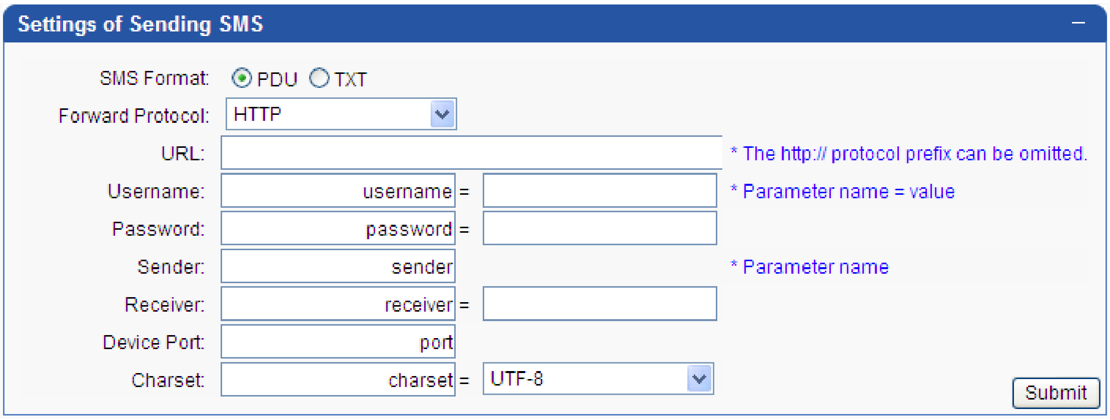


Fields are specified as following:

- Forward Protocol: Specify forward protocol sip.

- Server IP: Specify the sms server ip.
- Content-Type: Specify the Content-Type.
- Content Charset: Specify the content charset.

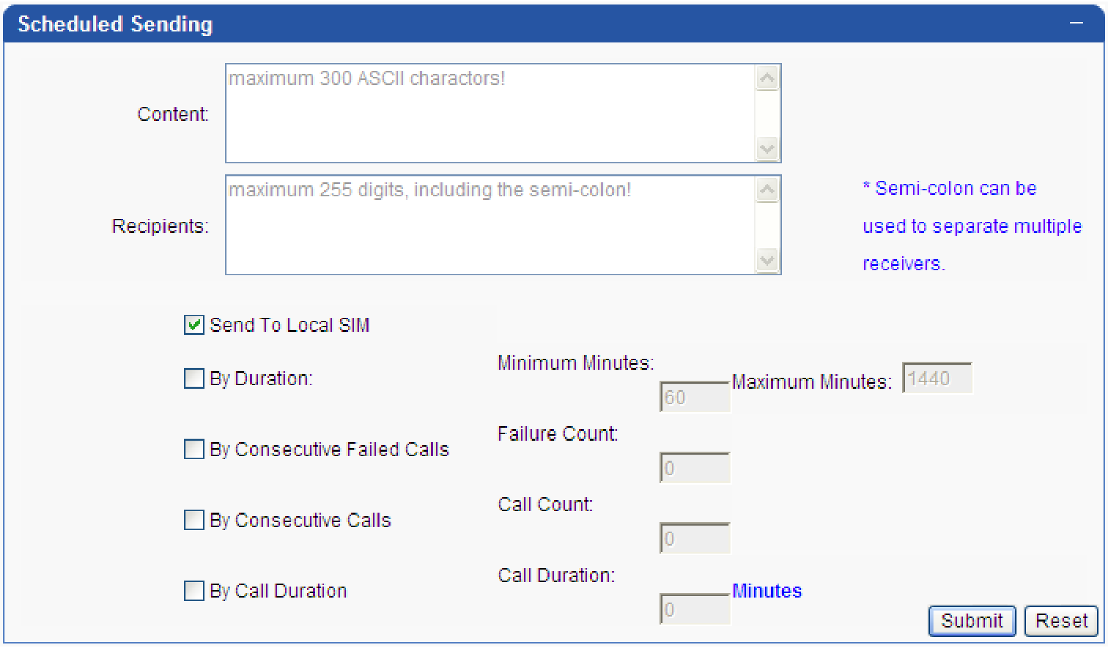
The screenshot below shows the operation mode to set http protocol of forwarding sms.



Fields are specified as following:

- Forward Protocol: Specify the forward protocol http.
- URL: Specify the sms server URL.
- Username: Specify the username.
- Password: Specify the password.
- Sender: Specify the sender.
- Receiver: Specify the receiver.
- Device Port: Specify the device port.
- Charset: Specify the charset UTF-8.

The screenshot below shows the operation mode to set scheduled sending sms

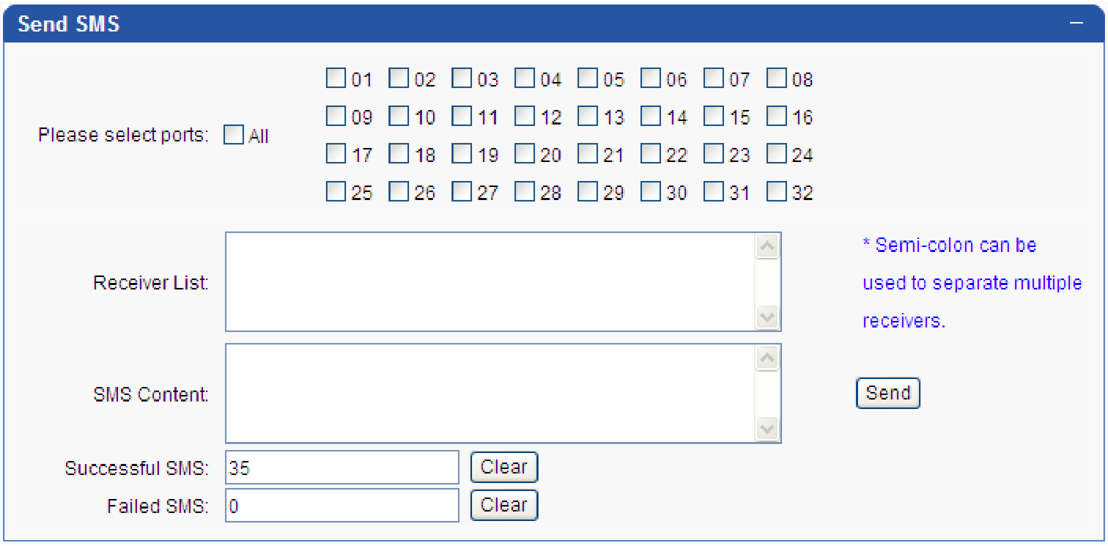


Fields are specified as following:

- Content: Specify the sms content.
- Recipients: Specify the recipients. Semi-colon can be used to separate multiple receivers.

- Send To Local SIM:Enable this feature and set the local SIM's number,the inter port will send sms.
- By Duration: Gateway will start sms sending by the device online time, and the time between minimum minutes and maximum minutes.
- By Consecutive Failed Calls:Gateway will start sms sending by consecutive failed calls.
- By Consecutive Calls:Gateway will start sms sending by consecutive calls.
- By Call Duration:Gateway will start sms sending by call duration.

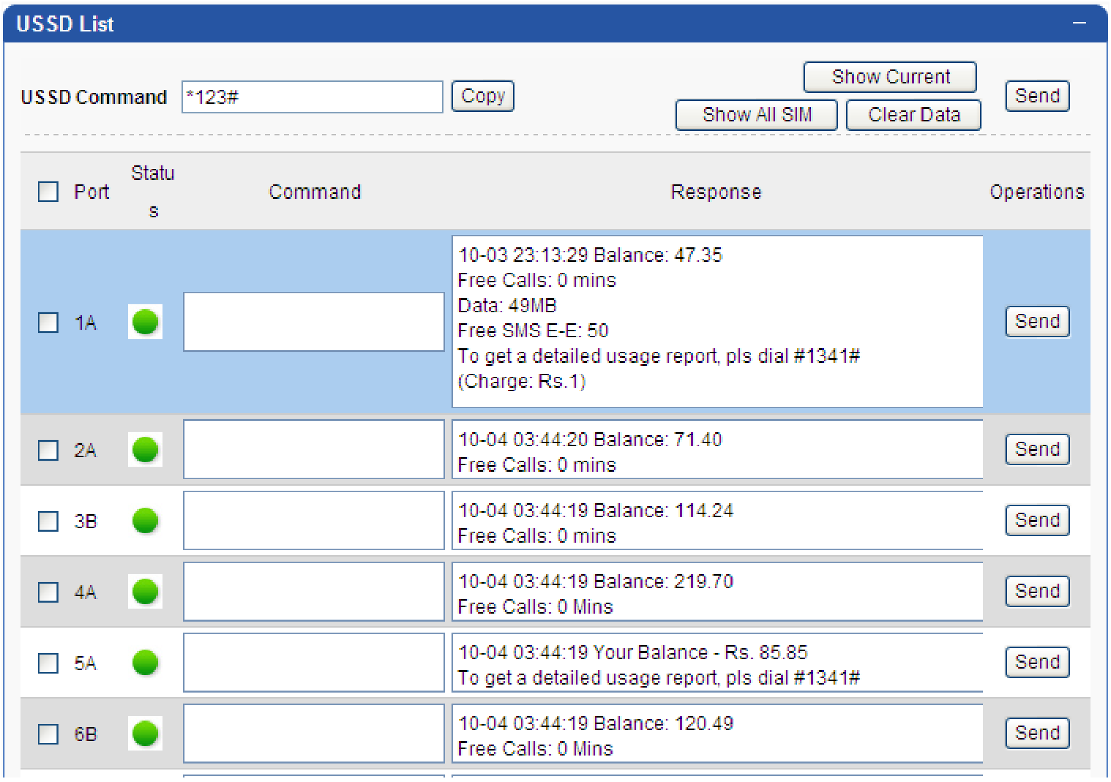
The screenshot below shows the operation mode to send SMS through the GoIP Gateway.



- Select port. The module here means GoIP mobile port and the SMS is sent out through the card which is in service on this port.
- Input the receivers separated by semi-colon.
- Input SMS content and click button send to send out the SMS.
- Field Received SMS is used to display the last response of the SMS sent out, if the response is not empty.
- Field Successful SMS Number records down the total number of SMS which is successfully sent out. Field Failed SMS Number records down the total number of SMS which is sent failed.

7.10 USSD Setting

The screenshot below shows the operation mode to send USSD through the GoIP Gateway.

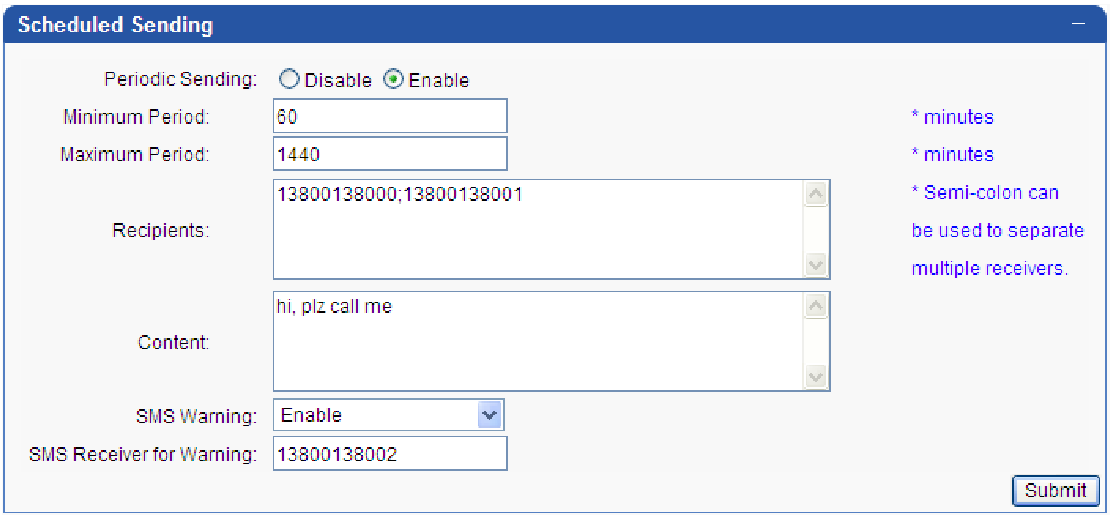


Fields are specified as following:

- USSD Command: The value of the USSD
- Port: Select tick to need send USSD ports
- Response: Show respond to the content of the carrier
- Send: Press this button, will start sending USSD

7.11 Automation

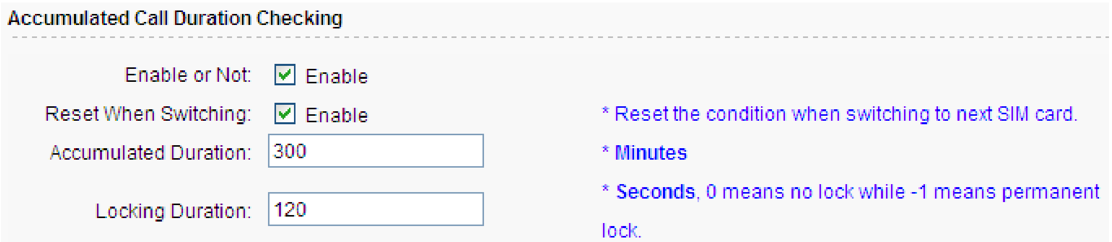
7.11.1 Scheduled Sending sms



Some mobile operators detect SIM cards used only in calling without sending SMS, will blocked SIM's.

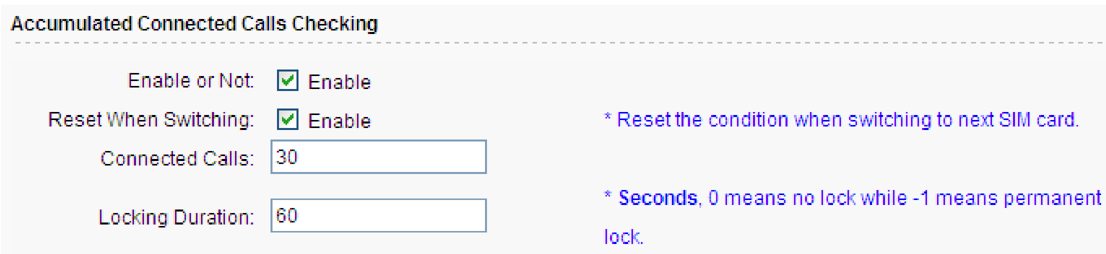
SMS Warning: A SMS warn the gateway manager to check the SIMs when they are locked(Not mobile operator blocking, it's the politic schedule to limit the SIM use time, use frequency to avoid blocking).

7.11.2 Accumulated Call Duration Checking



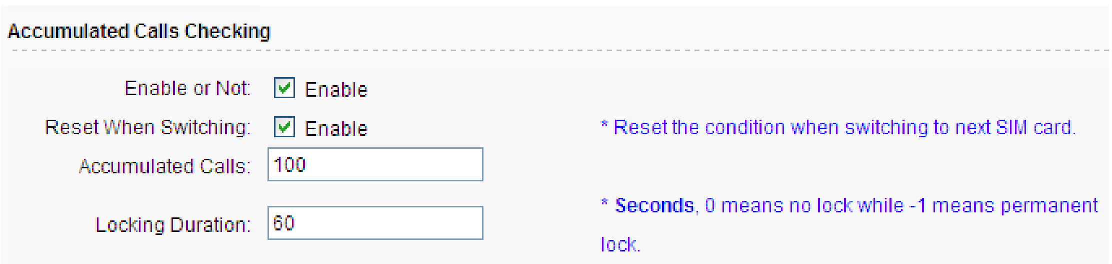
According to the actual situation, input the reasonable numerical. When the value is conform to the set, will automatically switches to the next sim card

7.11.3 Accumulated Connected Calls Checking



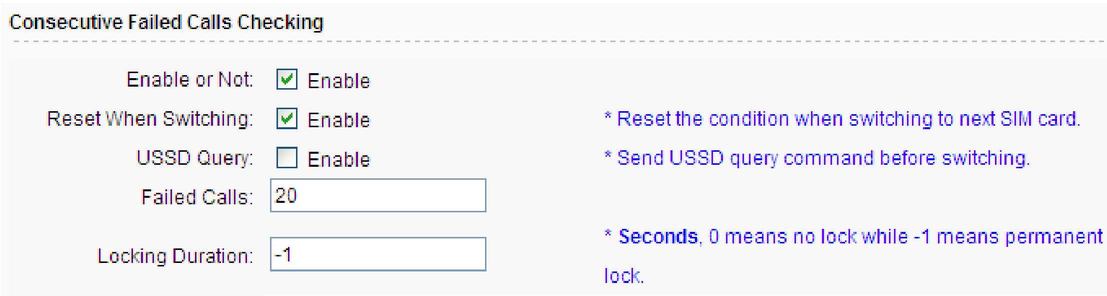
According to the actual situation, input the reasonable numerical. When the value is conform to the set, will automatically switches to the next sim card

7.11.4 Accumulated Calls Checking



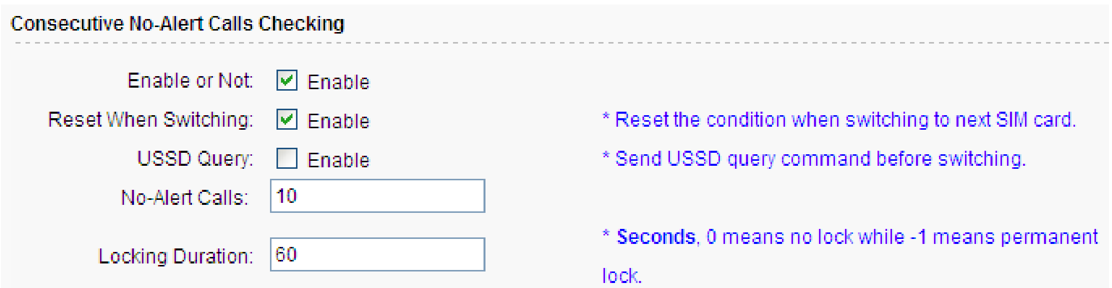
According to the actual situation, input the reasonable numerical. When the value is conform to the set, will automatically switches to the next sim card

7.11.5 Consecutive Failed Calls Checking



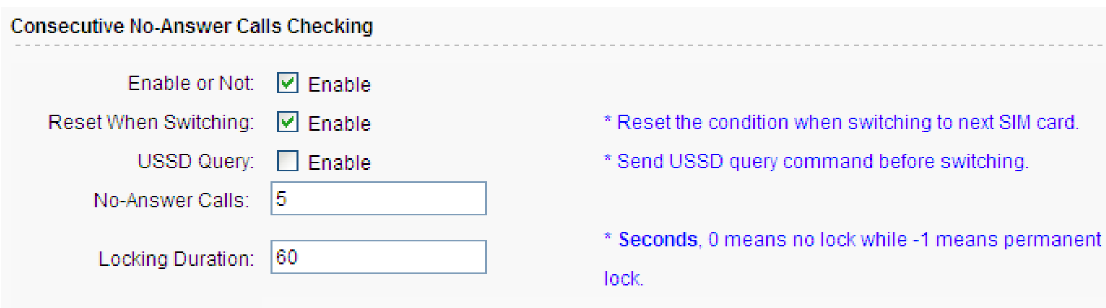
According to the actual situation, input the reasonable numerical. When the value is conform to the set, will automatically switches to the next sim card

7.11.6 Consecutive No-Alert Calls Checking



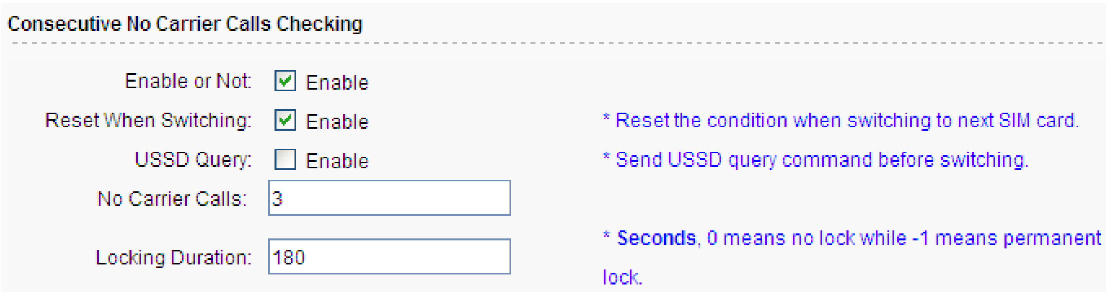
According to the actual situation, input the reasonable numerical. When the value is conform to the set, will automatically switches to the next sim card

7.11.7 Consecutive No-Answer Calls Checking



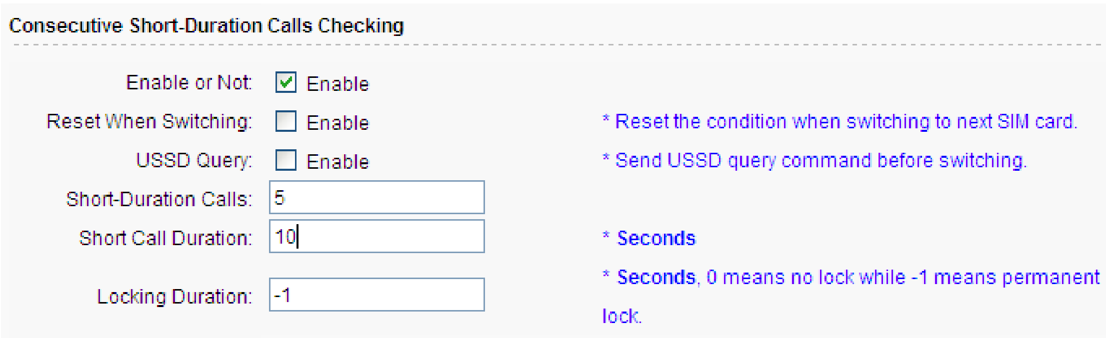
According to the actual situation, input the reasonable numerical. When the value is conform to the set, will automatically switches to the next sim card

7.11.8 Consecutive No Carrier Calls Checking



According to the actual situation, input the reasonable numerical. When the value is conform to the set, will automatically switches to the next sim card

7.11.9 Consecutive Short-Duration Calls Checking

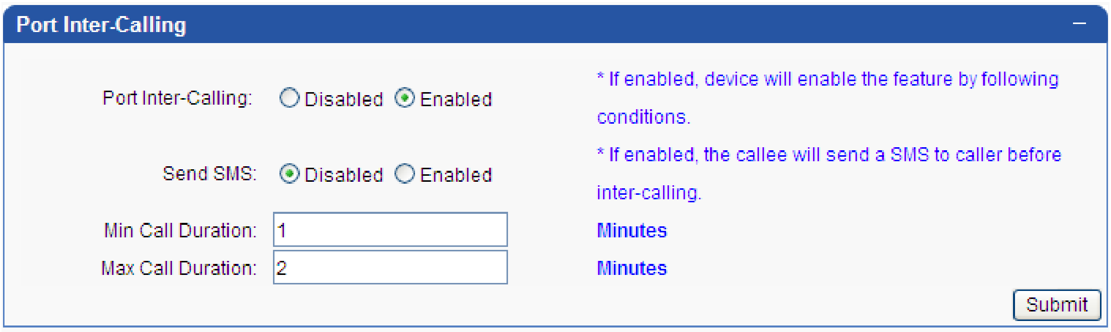


According to the actual situation, input the reasonable numerical. When the value is conform to the set, will automatically switches to the next sim card

7.12 InterCall Setting

7.12.1 Port Inter-Calling

The screenshot below shows the operation mode to set globally for port inter- calling.



Fields are specified as following:

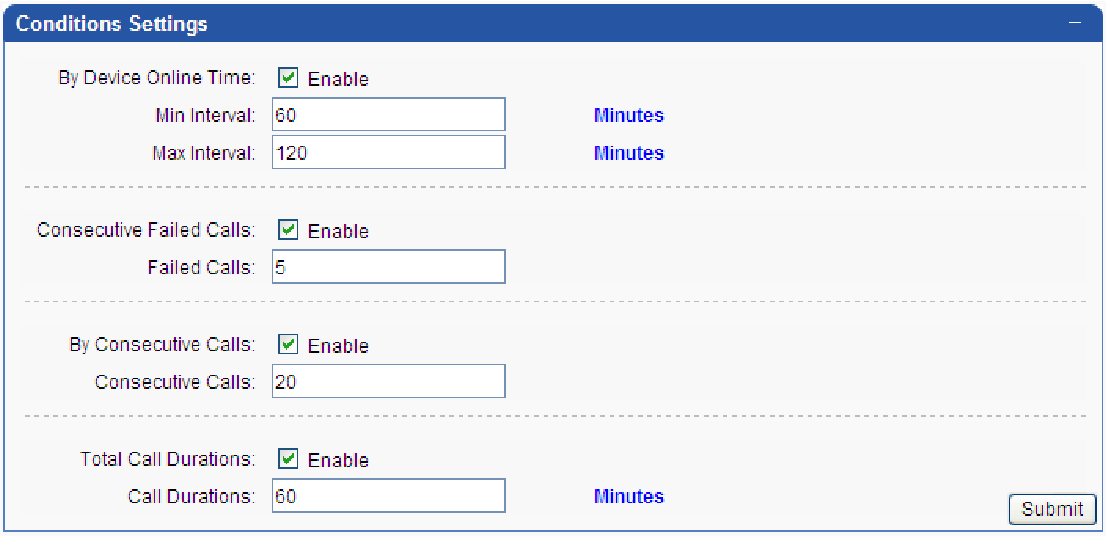
- Port Inter-Calling: Specify whether enable port inter-calling.
- Min Call Duration: Specify the minimum call duration.
- Max Call Duration: Specify the maximum call duration.

This panel allow SIMs in the gateway to call each other randomly. Consider that SIMs inside only call out all the time, so it's easy to be judged as an illegal use.

When enable "Port Inter-Calling", every SIM can receive income call in period which is custom option in "Conditions Settings".

7.12.2 Conditions Settings

The screenshot below shows the operation mode to set conditions settings of port inter-calling.



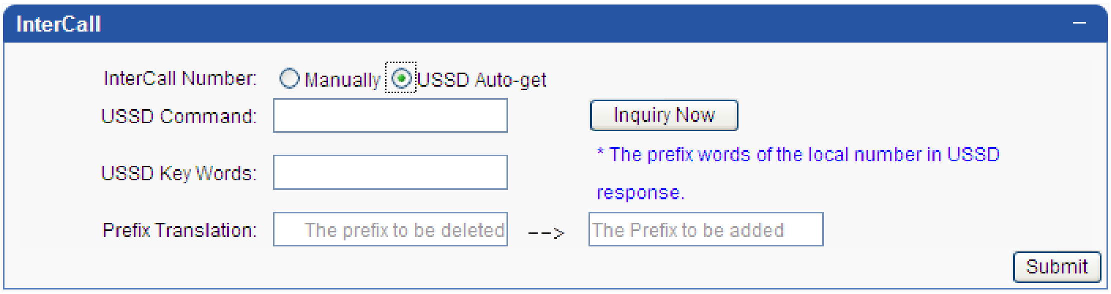
Fields are specified as following:

- By Device Online Time: Gateway will start port inter calling by the device online time, and the time between min interval and max interval.
- Consecutive Failed Calls: Gateway will start port inter calling when reaches the consecutive failed calls.
- By Consecutive: Gateway will start port inter calling when reaches the consecutive calls.
- Total Call Duration: Gateway will start port inter calling when reaches the call duration.

According to the actual situation, input the reasonable numerical.

7.12.3 SIM Num Settings

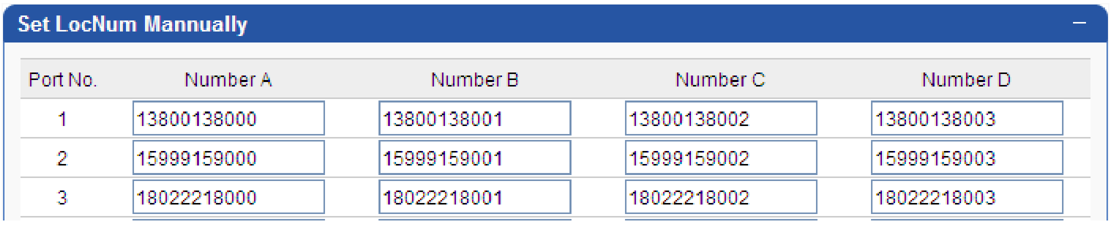
The screenshot below shows the operation mode to get Local Number by USSD



Fields are specified as following:

- USSD Auto-get: when enable it, gateway will get the sim number by ussd command
- USSD Command: Specify the ussd command for querying sim number.
- USSD Key Words: to pick the sim number.
- Prefix Translation: change the sim number prefix.

The screenshot below shows the operation mode to set local number manually.

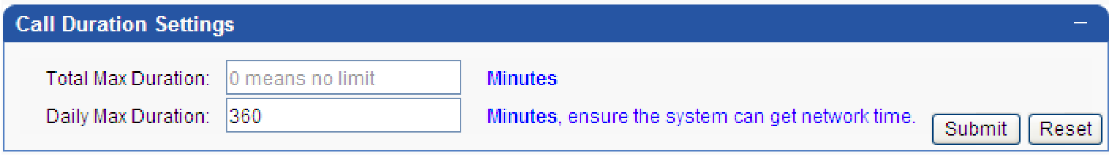


Fields are specified as following:

- Port No: The GoIP Gateway mobile port. Each port contains one or four card slots. Port No starts from 1 to 32.
- Number A: Specify the number for card A of the port
- Number B: Specify the number for card B of the port
- Number C: Specify the number for card C of the port
- Number D: Specify the number for card D of the port

7.13 Call Duration Settings

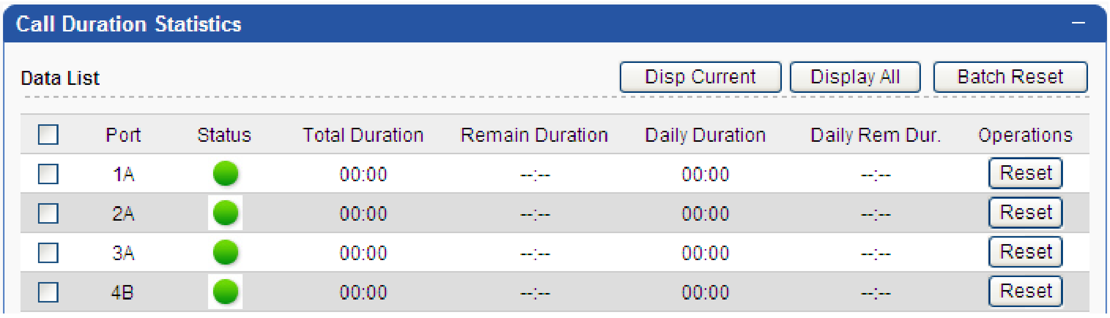
The screenshot below shows the call duration settings.



Fields are specified as following:

- Total Max Duration: once call duration reach total max duration, gateway will lock the sim.0 means no limit.
- Daily Max Duration: daily max duration need set the correct time. For example ,set 360 minutes, the sim has 360 minutes call duration everyday.

The screenshot below shows the call duration statistics.


Call Duration Statistics

Data List — Disp Current | Display All | Batch Reset

	Port	Status	Total Duration	Remain Duration	Daily Duration	Daily Rem Dur.	Operations
☐	1A	●	00:00	--:--	00:00	--:--	Reset
☐	2A	●	00:00	--:--	00:00	--:--	Reset
☐	3A	●	00:00	--:--	00:00	--:--	Reset
☐	4B	●	00:00	--:--	00:00	--:--	Reset

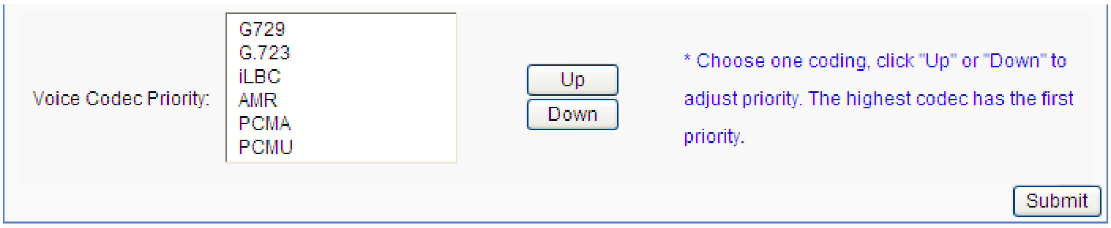
The columns are specified as following:

- Port: The GoIP Gateway mobile port. Each port contains one or four card slots. Port No starts from 1 to 32.
- Status: it shows the sim status.
- Total Duration: Specify the total duration has used.
- Remain Duration: Specify the remain duration.
- Daily Duration: Specify the daily duration has used.
- Daily Rem Dur.: Specify the daily remain duration.

❖8 System Setting

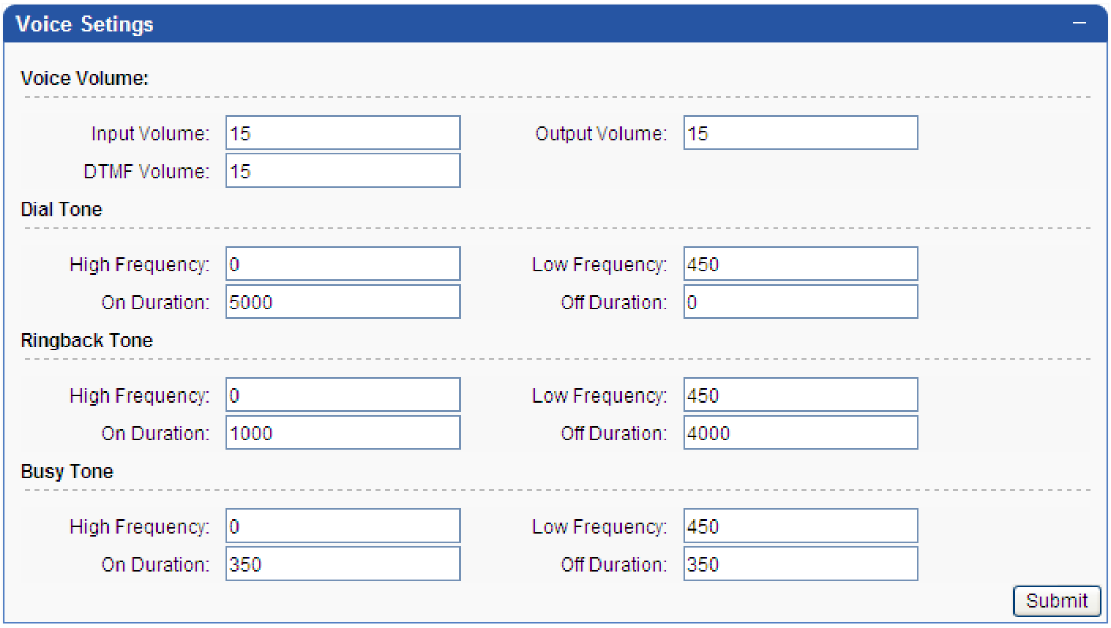
8.1 Voice and Codec

The screenshot below shows the operation mode to set codec priority.



Three codec types are provided to adjust GoIP Gateway to different network environment. It support G729a/b/e,G723.1,G711A/U law,iLBC,AMR auto- selecting.

The screenshot below shows the operation mode to set voice settings.



Voice Volume is used to specify the input voice volume, output voice volume andDTMF tone volume. The acceptable value for volume is an integer no less than 10 and no greater than 40.

The Dial Tone is sent to a customer or operator to indicate that the receiving end is ready to receive dial pulses or DTMF signals. It is used in all types of dial offices when the customer's or operator's dials produce dial pulses.

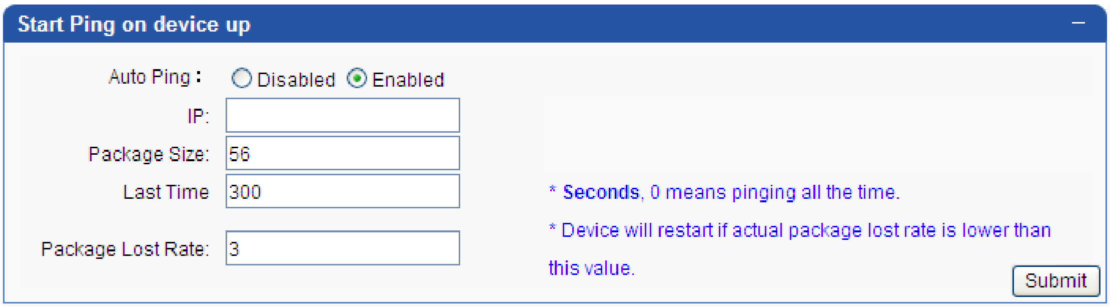
A Ring Back tone (or ringing tone) is an audible indication that is heard on the telephone line by the caller while the phone they are calling is being rung. It is normally a repeated tone, designed to assure the calling party that the called party's line is ringing.

The Busy Tone indicates that the called customer's line has been reached but that it is busy, being wrong, or on permanent signal. W hen an operator applies a busy signal, it is sometimes called a busy-back tone. Line Busy Tone is a Low Tone that is on and off every 0.5 second.

The settings of Dial Tone, Ring Back Tone and Busy Tone depend on area. The default settings for Asia are shown in the screenshot above for reference.

8.2 Network Debug

The screenshot below shows the auto ping settings

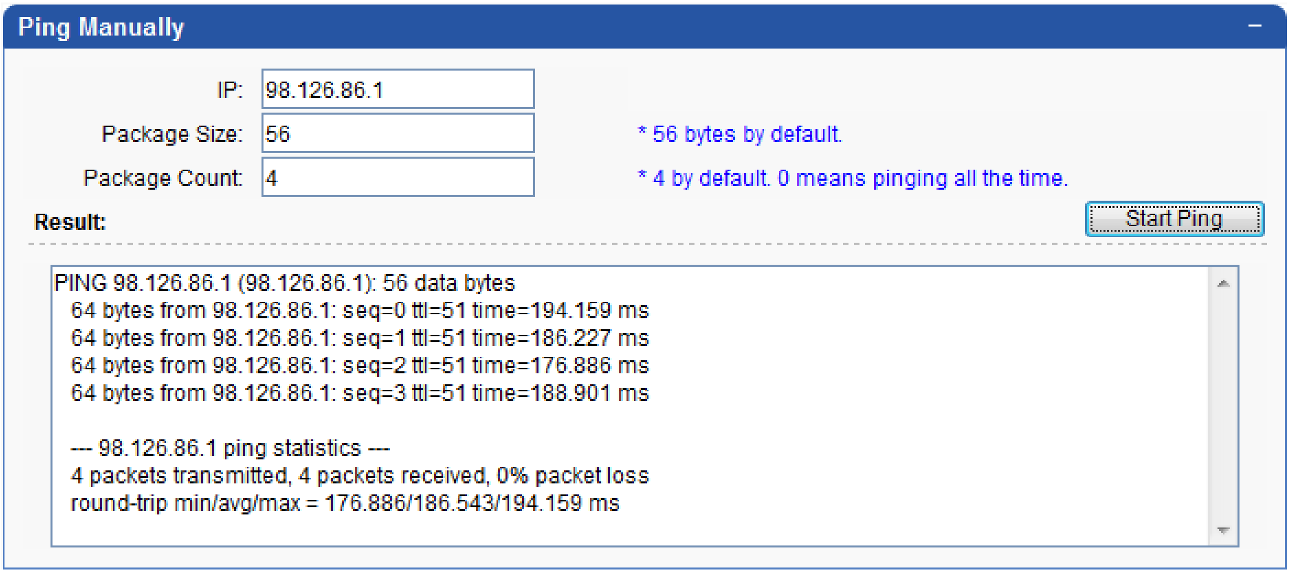


Fields are specified as following:

- Auto Ping: Specify whether enable auto ping, when the device power on to runing.
- IP Address: Specify the ip address.

- Packet Size: Specify the packet size.
- Last Time: Specify the ping duration.
- Package Loss Rate: Specify the package loss rate.

The screenshot below shows the manual ping settings



Fields are specified as following:

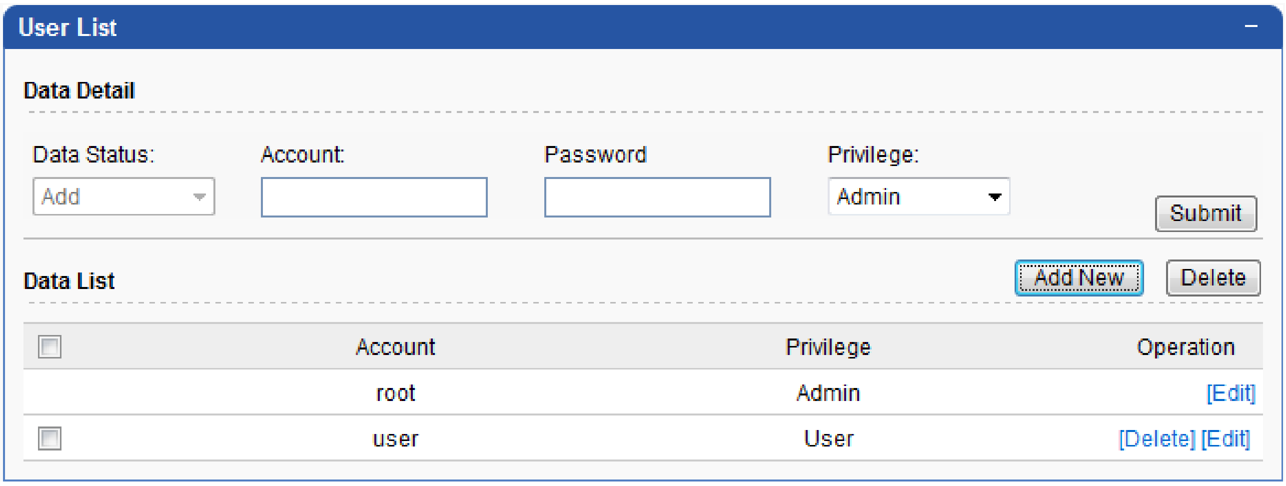
- IP Address: Specify the ip address.
- Packet Size: Specify the packet size.
- Packet Count: Specify the packet count.

The ping tool is easy to check the gateway network status. Especially when calls can't connect but every SIP parameters are correct, this tool will be helpful to find out problems.

8.3 User & Device

8.3.1 User

The screenshot below shows the operation mode to manage system user.



Default User

The default system user account is root. This account can't be deleted and only Password and Privilege can be modified for this account.

Add User

Click button Add New to expand the data input area to add new data. Fields are specified as following:

- Data status: Mark the status of current data record. Option values are Add/Edit. Value Add means the data is new while value Edit means the data is old.
- Account: The user account used to login web system. The account value can not be modified after save.
- Password: The password used to login web system.
- Privilege: The privilege of user. Option values are Admin/User.

Click button Submit on the right to save the new data record.

Edit User

All the user records are displayed in list. Two operations are provided on the right of each record.

Click Edit to expand the current data record to Data Detail Area which is above the Data List.

Click button Submit on the right to save the old data record.

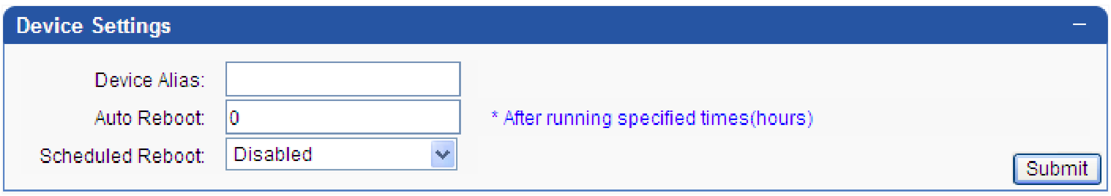
Delete User

Click Delete on the right of each record to delete the current record. A message box will be popped for delete confirmation.

Another shortcut button is also provided on the top right of Data List to delete multiple selected records in batch. A message box will be popped for confirmation of batch delete.

8.3.2 Device Settings

The screenshot below shows the operation mode to set Device settings.



Fields are specified as following:

- Device Alias: Specify the device alias.
- Auto Reboot: Specify the auto reboot time.
- Scheduled Reboot: Specify the scheduled reboot time.

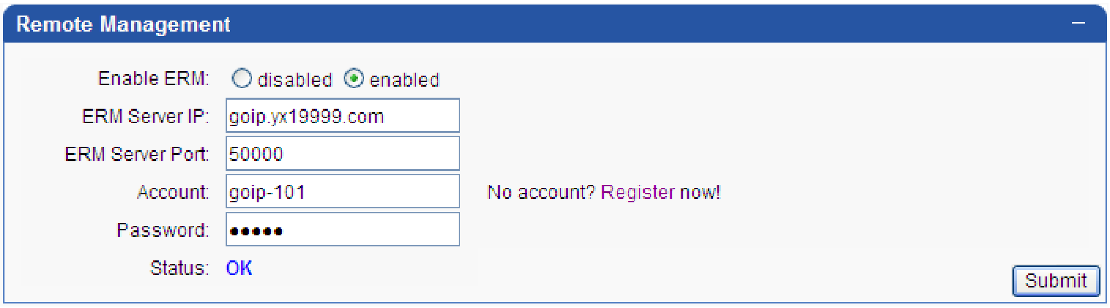
8.3.3 Date And Time

The screenshot below shows the operation of date and time settings.

The default time zone is UTC+8, you can change the time zone as your country. For example, Bangladesh is UTC+6, and change as +6. If your device is not touch with the internet and want to get accurate time, the time server will help.

8.3.4 Remote Mangement

The screenshot below shows the operation mode for remote management.



Remote Management is used to manage the GoIP Gateways located in other physical locations. Network must be available for the gateway to communicate with ERM Server.

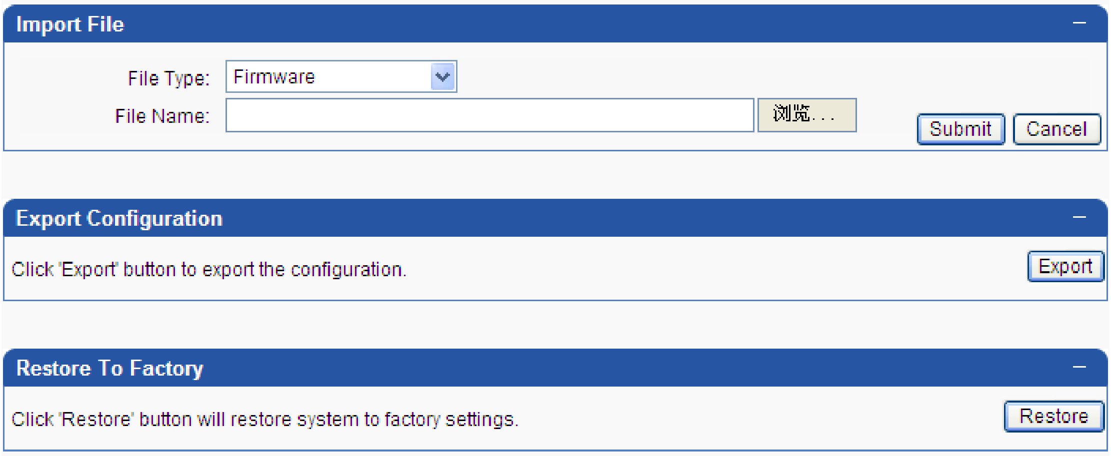
If ERM is enabled and correctly set, the GoIP will register to ERM server and set up the connection between itself and ERM server. Administrator can login ERM server and monitor all the registered GoIP Gateways. Commands can also be sent from ETMS server to certain gateway for management.

The configuration fields are specified as following:

- Enable ERM: Specify whether enable ETMS registration or not. Option values are Enabled/Disabled.
- ERM Server IP: Specify the ERM Server address.
- ERM Server Port: Specify the ERM server port.
- Account: Specify the account which create in the ERM.
- Password: Specify the password which create in the ERM.
- Status: Specify the registration status.

8.4 Update/ backup & Restore

The screenshot below shows the operation mode for system update/restore.



System Update

The content for system update includes:

- firmware
- configuration
- ramfs
- kernel
- uboot
- debug tools
- voice prompt
- voice cfg
- mac file
- lic file
- customized

The configuration fields are specified as following:

- File Type: Specify the content to update. Option values are listed above.
- File Name: Specify the content file name. Click button Browser and then select the target file from the popped file selection window.

Export Configuration

Click 'Export' button to export the configuration

Restore To Factory

System restore is used to restore the system to default settings. A message box will be popped for the confirmation of restore.

8.5 Save & Reboot

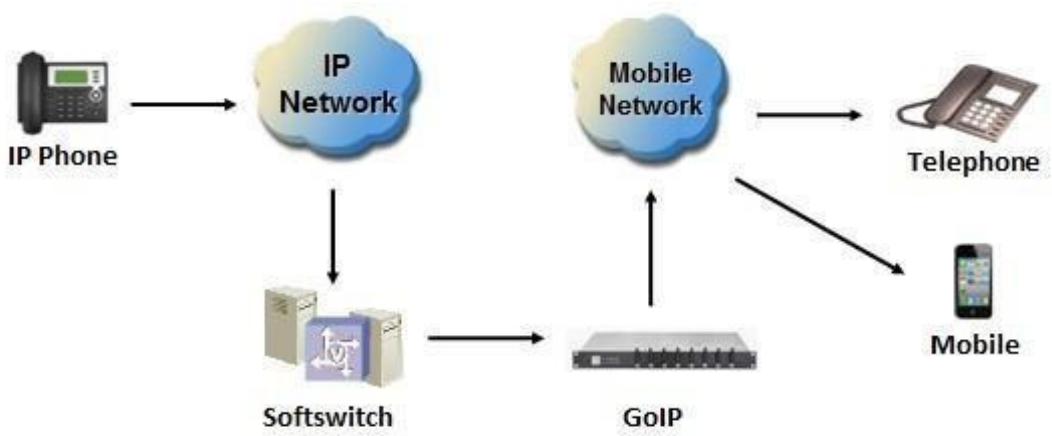
Generally, any modification should require the reboot of GoIP Gateway to bring the modification into effect. However, single Save without Reboot is also frequently used to save the modifications which will be effective on next reboot of GoIP Gateway.

The screenshot above shows the operation buttons. Button Save is used to save all the modifications while button reboot is used to save modifications first and then reboot device immediately.

❖9 Typical Used Scenario

This chapter presents some typical used scenarios for reference.

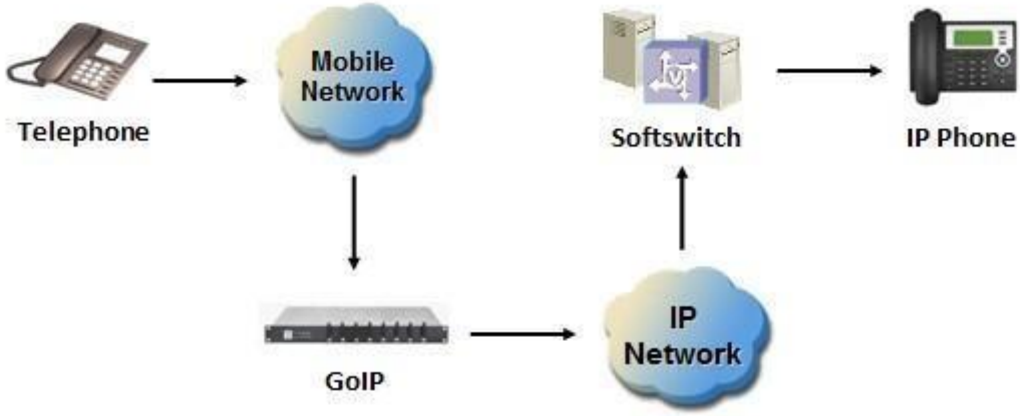
9.1 Landing from IP to Mobile Network



GoIP Gateway is now used more and more for telephone carriers to land their IP calls to mobile network. It plays the role of converting IP telephone signal to GSM telephone signal, relaying the media stream between IP network and Mobile network.

GoIP Gateway can be placed either in the LAN of Softswitch server or in public network environment which can be accessed by Softswitch server through public IP in different physical location.

9.2 Access from Mobile Network to IP



GoIP Gateway can be used as the access from mobile network to IP. Any call made to the mobile card inserted into GoIP Gateway will be routed to IP network and connected to Softswitch server. The Softswitch server can redirect the caller to final destination user.